IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

APPELLANT:	Eghart Fischer	GROUP ART UNIT: 2615
SERIAL NO.:	10/808,941	EXAMINER: John Douglas Suthers
FILED:	March 25, 2004	CONFIRMATION NO.: 6877
TITLE:	METHOD AND APPARATUS FOR IDENTIFYING THE DIRECTION OF INCIDENCE OF AN INCOMING AUDIO SIGNAL	

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APPELLANT'S MAIN APPEAL BRIEF

S I R:

In accordance with the provisions of 37 C.F.R. §41.37, Appellant herewith submits his main brief in support of the appeal of the above-referenced application.

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TABLE OF AUTHORITIES

Federal Cases

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REAL PARTY IN INTEREST

The real party in interest is the assignee of the application, Siemens Audiologische Tecknik, GmbH, a German company.

RELATED APPEALS AND INTERFERENCES

There are no related appeals and no related interferences.

STATUS OF CLAIMS

Claims 1-27 are the subject of the present appeal, and constitute all pending claims of the application. No claim was added or cancelled during prosecution before the Examiner.

Each of claims 1-27 stands as being finally rejected, in the July 17, 2008 Office Action.

STATUS OF AMENDMENTS

Amendment "B" under 37 C.F.R. §1.116 was filed on October 14, 2008, to correct a typographical error in claim 1. In an Advisory Action dated November 6, 2008, Amendment "B" was stated to be entered and to overcome the objection to claim 1, but the Amendment was stated not to place the application in condition for allowance because the prior art rejections were maintained.

SUMMARY OF CLAIMED SUBJECT MATTER

Independent claims 1 and 23 (the only independent claims on appeal) are set forth below with exemplary citations to the drawings and specification identifying disclosure locations for each of the relevant limitations of those claims.

1. A method for determining the direction of incidence of an incoming audio signal from an acoustic source to a directional microphone system, having at least two microphones (Fig. 1, directional microphone system RM1 with two

microphones M1, M2 and/or directional microphone system RM2 with three microphones M3, M4, M5; p.10, l.8-15), comprising the steps of:

detecting said incoming audio signal with said at least two microphones and, in each of said at least two microphones, producing an output microphone signal therefrom (p.11, l.21 - p.12, l.5);

generating at least two phase-shifted directional microphone signals by phase shifting at least one output microphone signal relative to another output microphone signal (p.12, l.22 - p.12, l.4; phase shifter PH in Fig. 2) and combining the phase-shifted output microphone signals with respective weightings (p.14, l.18-20), the respective weightings defining a direction-dependent sensitivity distribution, having a minimum in one direction, for the respective directional microphone signals (Fig. 3; p.14, l.1-14, p. 14, l.24-p.15, l.8);

assessing each of said directional microphone signals with respect to a quantity that indicates an influence, on the respective directional microphone signal, by the associated direction-dependent sensitivity distribution (assessment unit B in Fig. 4; p.15, l.9-13); and

comparing the respective quantities of the respective directional microphone signals with each other to identify a quantity having an extreme value (p.15, l.15-20), and determining the direction of incidence of said incoming audio signal as being the direction at which the minimum of the direction-dependent sensitivity distribution for the directional microphone signal having said extreme value is located (p.15, l.20-22).

23. An apparatus for determining a direction of incidence of an incoming audio signal comprising:

a directional microphone system having at least two microphones for detecting said incoming audio signal (Fig. 1, directional microphone system RM1 with two microphones M1, M2 and/or directional microphone system RM2 with three microphones M3, M4, M5; p.10, l.8-15), each of said at least two microphones generating a microphone signal therefrom (p.11, l.21 - p.12, l.5);

a phase-shifter that phase-shifts at least one microphone signal of said system relative to another microphone signal of said system (p.12, l.22 - p.12, l.4; phase shifter PH in Fig. 2);

weighting units (G1, G2, G3, G4 in Fig. 4) for respectively weighting said phase-shifted microphone signals for producing at least two directional microphone signals, the respective weightings (p.14, l.18-20) defining a direction-dependent sensitivity distribution for each of said directional microphone signals (Fig. 3; p.14, l.1-14, p. 14, l.24-p.15, l.8);

an assessment unit for assessing the respective directional microphone signals with respect to a quantity representing an influence of the direction-dependent sensitivity distribution on the directional microphone signal (assessment unit B in Fig. 4; p.15, l.9-13); and

a determination unit (assessment unit B) that identifies one of said directional microphone signals having an extreme value of said quantity compared (in assessment B) to the other directional microphone signals, and for

determining the direction of incidence of said incoming audio signal as being a direction in which a minimum of the direction-dependent sensitivity distribution of said one of said directional microphone signals is located (p.15, l.15-22).

Copies of Figures 1, 2, 3 and 4 as originally filed are submitted herewith as Exhibit "A".

GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

Whether the subject matter of claims 1, 2, 5-8, 11, 12 and 15-23 is anticipated under 35 U.S.C. §102(b) by United States Patent No. 6,069,961 (Nakazawa, Exhibit "B");

Whether the subject matter of claims 3, 4, 9 and 10 would have been obvious to a person of ordinary skill in the field of designing hearing-assist devices, under the provisions of 37 C.F.R. § 103(a), based on the teaching of Nakazawa; and

Whether the subject matter of claims 13, 14 and 24-27 would have been obvious to a person of ordinary skill in the field of designing hearing-assist devices, under the provisions of 35 U.S.C. §103(a), based on the teachings of Nakazawa in view of United States Patent No. 6,584,203 (Elko et al., Exhibit "C")

ARGUMENT

Anticipation Rejection of Claims 1, 2, 5-8, 11, 12 and 15-23 by Nakazawa

In substantiating the anticipation rejection based on Nakazawa, Appellant respectfully submits the Examiner has ignored important claim language in independent claims 1 and 23. The Examiner stated the Nakazawa reference discloses providing at least two directional microphone signals with respective weightings, by virtue of the subtractors 11A weighting one signal in the Nakazawa

circuit as a positive one, and the other as a minus one. As to the remainder of this step in claim 1 and the corresponding feature of claim 23, namely that the weightings define a direction-dependent sensitivity distribution having a minimum in one direction, for the respective directional microphone signals, the Examiner merely cited Figure 1B of Nakazawa. Figure 1B of Nakazawa, however, does not illustrate a direction-dependent sensitivity *distribution,* but merely illustrates the situation that occurs when subtraction of the signals from two microphones is undertaken. This is also why the assignment of minus one to one of the directional microphone signals is not a "weighting" in the sense set forth in claim 1 of the present application, since it is not "sensitive" to any direction, but is simply an arbitrary assignment of a polarity that is given to one of the incoming signals. Since every signal from the microphone that is connected to subtractor 11A in each block for each pair of microphones *always* has the *identical* value (namely minus one) assigned thereto, it is clear that this value is not dependent on anything, and is thus not sensitive to any direction, and thus does not represent any type of distribution. Moreover, since that "weighting" (if it is a weighting at all) is constant, i.e. it never changes, it clearly does not have a minimum in one direction, as also explicitly set forth in claims 1 and 23.

This weighting of the respective directional signals with a value that defines a direction-dependent sensitivity distribution is important to each of the subsequent steps in claim 1 on appeal, namely the "assessing" step and the "comparing" step. The signals weighted in the manner set forth in the first step of claim 1 and the corresponding feature of claim 23 are the signals that are assessed and compared in those subsequent steps, and since the Nakazawa reference does not disclose a weighting as set forth in this language of claims 1 and 23, even if the Nakazawa

reference discloses some type of “assessing” and/or some type of “comparing” , those steps of Nakazawa are not and cannot be comparable to the “assessing” and “comparing” steps of claim 1 nor the corresponding features of claim 23, because those operations in Nakazawa do not act on signals that have been weighted in the manner set forth in claims 1 and 23 of the present application.

In response to these arguments, the Examiner in the Final Rejection stated such a disclosure is present in the Nakazawa reference by virtue of the subtractors 11A weighting one signal in the Nakazawa circuit as a positive one, and the other as a minus one. This assignment of mathematical signs is not a trivial difference between the subject matter disclosed and claimed in the present application and the subject matter disclosed in the Nakazawa reference, because, as noted above, it is further stated in the independent claims of the present application that the aforementioned weightings define a direction-dependent sensitivity distribution that has a minimum in one direction, for the respective directional microphone signals. The Examiner stated that the Nakazawa reference provides such a disclosure in Figure 1B thereof, but Appellant submits that Figure 1B of Nakazawa does not illustrate any type of direction-dependent sensitivity *distribution*, but merely illustrates the situation that occurs when subtraction of the signals from two microphones is undertaken. Therefore, the assignment of a minus one to one of the directional microphone signals is not a “weighting” in the sense of claims 1 and 23, since it is not “sensitive” to any direction, but is simply an arbitrary assignment of a polarity that is given to one of the incoming signals. Since *every* signal from the microphone that is connected to the subtractor 11A in each block for each pair of microphones *always* has the *identical* value (namely minus one) assigned thereto,

this value is not and cannot be dependent on anything, and thus is not "sensitive" to any direction, and therefore does not represent any type of "distribution." Moreover, since the "weighting" that the Examiner has found to be disclosed in the Nakazawa reference always has the same value, i.e., it never changes, it clearly does not have a minimum in one direction, as also explicitly required in the independent claims.

The aforementioned weighting allows the subsequent "assessing" and "comparing" that is set forth in each of the independent claims of the present application to take place. If such weighting is not present, as in the Nakazawa reference, the "assessing" and "comparing" steps cannot proceed in Nakazawa in the same manner as set forth in claim 1 and the corresponding features of claim 23, because the results of the "assessing" and "comparing" steps are inextricably tied to the fact that the signals that are being assessed and/or the signals being compared do, in fact, have the aforementioned "weightings." In response to these previously-made arguments, the Examiner, in the July 17, 2008 Final Office Action, the Examiner stated that the directional characteristics shown in Figure 1 of the Nakazawa reference are "direction-dependent sensitivity distributions," that exhibit sensitivity to sound in a given direction. The Examiner stated the weightings provided by the subtractors 11A, and the choice of microphone inputs, define the pattern. The Examiner stated "Appellant in general argues points about the weighting that are claimed to belong to the direction-dependent sensitivity distribution, not the weightings." This statement by the Examiner is not fully understood since each of the independent claims of the present application explicitly states that the respective weightings themselves define the direction-dependent sensitivity distribution. In the claims of the present application, if the respective

weightings were not present (and applied to the phase-shifted output microphone signals) there would no sensitivity distribution, since the sensitivity distribution in the subject matter of the present application arises due to the weightings themselves. Even if the assignment of positive one and minus one is considered (contrary to Appellant's arguments above) to be a "weighting," there still would be (allegedly) a direction-dependent sensitivity distribution in the Nakazawa reference, even if such (alleged) "weightings" were not present. Such is not the case in the subject matter of the independent claims of the present application. Moreover, the Examiner stated that the distributions in Nakazawa are dependent on direction and have minimums as shown at angle zero in Figure 1B of the Nakazawa reference. As noted in Appellant's previous response, Figure 1B in Nakazawa does not show any type of *distribution*, but merely illustrates the result that occurs after subtraction of the signals from the two microphones. At page 3 of the Final Office Action, the Examiner argues that this subtractions represents a phase shift of 180°, but a person of ordinary skill in the field of microphone signal processing would clearly know that this is not the case in any realistic circuit. In order for subtraction of one signal from another to result in, or be the equivalent of, a phase shift of 180°, the special case would have to exist of a purely sinusoidal microphone signal, because only then would $\sin(\pi x) = - \sin(x)$. The assumption of purely sinusoidal microphone signals, however, is completely unrealistic, and would not be an assumption that a person of ordinary skill in the field of audio signal processing would make as to the signals that exist in any realistic audio processing circuit. Moreover, the claims of the present application do not claim a phase shift plus a subsequent addition, as taught by Nakazawa.

In the aforementioned Advisory Action, the Examiner made the following statement:

> Perhaps there is a difference in the interpretation of the claim by the Examiner to that intended by the applicant. The applicant argues that "the Nakazawa reference does not disclose providing at least two directional microphone signals with respective weightings". However, the claim language only claims "combining the phase-shifted output microphone signals with respective weightings". The Examiner's interpretation includes "generating at least two phase-shifted directional microphone signals", each by the same manner which includes the combining process as above.

In response, Appellant submits that in the language of claims 1 and 23, it is clear that the phrase "combining the phase-shifted output microphone signals with respective weightings" means that the phase-shifted output microphone signals are given respective weightings and the weighted, phase-shifted output microphone signals are then combined. This is clear because of the language immediately following the "combining" phrase in each of claims 1 and 23, explicitly stating that the respective weightings define a direction-dependent sensitivity distribution, having a minimum in one direction, *for the respective directional microphone signals*. Appellant submits that the plain meaning of this language is that each microphone signal is given a weighting that defines a direction-dependent sensitivity distribution for that particularly directional microphone signal. Appellant is unable to interpret this language in any other manner and, for the reasons discussed above, no weighting of the type described in this claim language is disclosed in the Nakazawa reference.

An anticipating reference must describe all of the elements and limitations of the claim in a single reference and enable one of skill in the field of the invention to make and use the claimed invention. *Merck & Co. v. Teva Pharmaceuticals USA,*

Inc., 347 F.3rd 1367, 1372, 68 U.S.P.Q. 2nd 185 (Fed. Cir. 2003). Anticipation under 35 U.S.C. §102 requires that a single prior art reference disclose each and every limitation of the claimed invention. *Moba, B. V. v. Diamond Automation, Inc.*, 325 F.3d 1306, 1321, 66 U.S.P.Q. 2d 1429 (Fed. Cir. 2003), *cert. denied*, 540 U.S. 982 (2003).

A single reference must describe the claimed invention with sufficient precision and detail to establish that the subject matter existed in the prior art. *Verve, LLC v. Crane Cams, Inc.*, 311 F.3d 1116, 1120, 65 U.S.P.Q. 2d 1051 (Fed. Cir. 2002). The reference must describe the Appellants' claimed invention sufficiently to have placed a person of ordinary skill in the field of the invention in possession of it. *In re Spada* 911 F.2d 705, 708, 15 U.S.P.Q. 2d 1655, 1657 (Fed. Cir. 1990). A single prior art reference anticipates a patent claim if it expressly or inherently describes each and every limitation set forth in the patent claim. *Trintec Industries, Inc. v. Top-U.S.A. Corp.*, 295 F.3d 1292, 1295, 63 U.S.P.Q. 2d 1597 (Fed. Cir. 2002).

For all of the above reasons, Appellant respectfully submits that neither of claims 1 or 23 is anticipated by Nakazawa. Claims 2, 5-8, 11, 12 and 15-22 add further method steps to the novel method of claim 1, and therefore none of those dependent claims is anticipated by Nakazawa, for the same reasons discussed above in connection with independent claim 1.

Obviousness Rejection of Claims 3, 4, 9 and 10 Based on Nakazawa

The above arguments with regard to the anticipation rejection of claim 1 are applicable to the obviousness rejection of claims 3, 4, 9 and 10 based on Nakazawa. The above arguments demonstrate that, not only is the subject matter of claims 3, 4,

9 and 10 (all of which embody the subject matter of claim 1 therein) not explicitly disclosed in the Nakazawa reference, but also that modification of the Nakazawa reference in order to arrive at the subject matter of those dependent claims would not have been obvious to a person of ordinary skill in the field of designing hearing-assist devices.

The Federal Circuit stated in *In re Lee* 227 F.3d 1338, 61 U.S.P.Q. 2d 1430 (Fed. Cir. 2002):

> "The factual inquiry whether to combine references must be thorough and searching. ...It must be based on objective evidence of record. This precedent has been reinforced in myriad decisions, and cannot be dispensed with."

Similarly, quoting *C.R. Bard, Inc. v. M3 Systems, Inc.*, 157 F.3d 1340, 1352, 48 U.S.P.Q. 2d 1225, 1232 (Fed. Cir. 1998), the Federal Circuit in *Brown & Williamson Tobacco Court v. Philip Morris, Inc.*, 229 F.3d 1120, 1124-1125, 56 U.S.P.Q. 2d 1456, 1459 (Fed. Cir. 2000) stated:

> [A] showing of a suggestion, teaching or motivation to combine the prior art references is an 'essential component of an obviousness holding'.

In *In re Dembiczak,* 175 F.3d 994,999, 50 U.S.P.Q. 2d 1614, 1617 (Fed. Cir. 1999) the Federal Circuit stated:

> Our case law makes clear that the best defense against the subtle but powerful attraction of a hindsight-based obviousness analysis is rigorous application of the requirement for a showing of the teaching or motivation to combine prior art references.

Consistently, in *In re Rouffet,* 149 F.3d 1350, 1359, 47 U.S.P.Q. 2d 1453, 1459 (Fed. Cir. 1998), the Federal Circuit stated:

> [E]ven when the level of skill in the art is high, the Board must identify specifically the principle, known to one of ordinary skill in the art, that suggests the claimed combination. In other words, the Board must explain the reasons one of ordinary skill in the art would have been

> motivated to select the references and to combine them to render the claimed invention obvious.

In *Winner International Royalty Corp. v. Wang,* 200 F.3d 1340, 1348-1349, 53 U.S.P.Q. 2d 1580, 1586 (Fed. Cir. 2000), the Federal Circuit stated:

> Although a reference need not expressly teach that the disclosure contained therein should be combined with another, ... the showing of combinability, in whatever form, must nevertheless be clear and particular.

Lastly, in *Crown Operations International, Ltd. v. Solutia, Inc.,* 289 F.3d 1367, 1376, 62 U.S.P.Q. 2d 1917 (Fed. Cir. 2002), the Federal Circuit stated:

> There must be a teaching or suggestion within the prior art, within the nature of the problem to be solved, or within the general knowledge of a person of ordinary skill in the field of the invention, to look to particular sources, to select particular elements, and to combine them as combined by the inventor.

Appellants submit that the decision of the United States Supreme Court in *KSR International Co. v. Teleflex Inc.,* _____U.S. _____, 127 S.Ct. 1727, 82 USPQ 2d 1385 (2007), and the United States Patent and Trademark Office guidelines for applying that decision, support the position of the Appellants. That decision, although stating that it is not always required to point to a specific teaching in a prior art reference in order to substantiate a rejection under 35 U.S.C. §103(a), by no means approved ignoring the above long-standing precedent, and certainly did not represent a blanket overruling of that precedent. In the *KSR* decision, the Supreme Court stated, *under certain circumstances*, it may not be necessary to point to a specific passage in a prior art reference as evidence of motivation, guidance or inducement in order to modify that reference in a manner that obviates the patent claim in question. The Supreme Court stated that if a person of ordinary skill in the

art can implement a *predictable variation* and would see the benefit of doing so, Section 103(a) likely bars patentability.

Nevertheless, the Supreme Court also stated that the requirement to find a teaching, suggestion or motivation in the prior art "captures a helpful insight." The Supreme Court stated that although common sense directs caution as to a patent application claiming as innovation the combination of two known devices according to their established functions, it can be important to identify a reason that would have prompted a person of ordinary skill in the art to combine the elements as the new invention does. The Supreme Court, however, stated that not every application requires such detailed reasoning. The Supreme Court stated that helpful insights need not become rigid and mandatory formulas. The Supreme Court only stated that if the requirement to find a teaching, suggestion or motivation is required in such a rigid, formulaic manner, it is then inconsistent with the precedence of the Supreme Court. In fact, the Supreme Court stated that since the "teaching, suggestion or motivation" test was devised, the Federal Circuit doubtless has applied it in accord with these principles in many cases. The Supreme Court stated there is no necessary inconsistency between this test and an analysis conducted under the standards of *Graham v. Deere*. The Supreme Court stated the only error is transforming this general principle into a "rigid rule limiting the obviousness inquiry."

Therefore, Appellants submit this decision of the Supreme Court does not in any manner approve, much less require, the absence of a rigorous evidentiary investigation on the part of the Examiner in order to substantiate most rejections under 35 U.S.C. §103(a). Only under the somewhat unusual, and very limited, circumstances outlined by the Supreme Court in the *KSR* decision might the

Supreme Court excuse the absence of such a rigorous evidentiary investigation in reaching a conclusion of obviousness under 35 U.S.C. §103(a).

This view of the *KSR* decision has been substantiated by the United States Court of Appeals for the Federal Circuit in *Takeda Chemical Industries Limited v. Alphapharm Pty.Ltd.*,492 F.3d 1350, 83 U.S.P.Q.2d, 169 (Fed. Cir. 2007), which was one of the earliest decisions of the Federal Circuit after the *KSR* decision was decided by the Supreme Court. The *Takeda* decision concerned a chemical patent that was the subject of an infringement lawsuit, and which was attacked by the infringer on the basis of the claimed subject matter being "obvious to try." After acknowledging that the *KSR* decision held that the teaching-suggestion-motivation test should not be applied rigidly, the Federal Circuit stated that the *KSR* decision actually recognized the value of that test in determining whether the prior art provided a *reason* for one of skill in the art to make the claimed combination. The Federal Circuit stated this is consistent with the Federal Circuit precedent in *In re Dillon,* 919 F.2d 688 (Fed. Cir. 1990) and in *In re Deuel*, 51 F.3d 1552 (Fed. Cir. 1995). The Federal Circuit stated that in cases involving new chemical compounds, it remains necessary to identify some reason that would have led a chemist to modify a known compound in a particular manner to establish *prima facie* obviousness of the new claimed compound. In the *Takeda* decision, the Federal Circuit stated:

> The *KSR* Court recognized that "[w]hen there is a design need or market pressure to solve a problem and there are a finite number of identified, predictable solutions, a person of ordinary skill has good reason to pursue the known options within his or her technical grasp," *KSR*, 127 S.Ct. at 1732. In such circumstances, "the fact that a combination was obvious to try might show that it would obvious under §103." *id.* that is not the case here. Rather than identify predictable solutions for antidiabetic treatment, the prior art disclosed a broad selection of compounds, any one of which could have been selected as a lead compound for further investigation.

Appellant therefore submits that the Examiner has failed to properly substantiate the rejection of claims 3, 4, 9 and 10 under 35 U.S.C. §103(a) as being unpatentable over Nakazawa.

Obviousness Rejection of Claims 13, 14 and 24-27 Based on Nakazawa and Elko et al.

The previous arguments with regard to independent claims 1 and 23 are equally applicable to the rejection of claims 13, 14 and 24-27 based on Nakazawa and Elko et al. Claims 13 and 14 add further steps to the novel method of claim 1, and claims 24-27 add further components to the novel combination of claim 23. Even if the Examiner's statements concerning the individual teachings of the Elko et al. reference are correct, modifying the Nakazawa reference in accordance with those teachings still would not result in the subject matter of any of claims 13, 14 or 24-27, which respectively embody the subject matter of the independent claims therein.

The above discussion of the relevant law for substantiating a rejection under 35 U.S.C. §103(a) is applicable to this rejection as well.

Appellant therefore submits that none of claims 13, 14 or 24-27 would have been obvious to a person of ordinary skill in the field of designing hearing-assist devices, under the provisions of 35 U.S.C. §103(a), based on the teachings of Nakazawa and Elko et al.

CONCLUSION

For the above reasons, Appellant respectfully submits that the rejection of claims 1-27 are in error in law and in fact, and reversal of those rejections is proper.

This Appeal Brief is accompanied by electronic payment for the requisite fee in the amount of $540.00.

Submitted by,

Steven H. Noll (Reg. 28,982)

SCHIFF, HARDIN LLP
CUSTOMER NO. 26574
Patent Department
6600 Sears Tower
233 South Wacker Drive
Chicago, Illinois 60606
Telephone: 312/258-5790
Attorneys for Appellant.

CLAIMS APPENDIX

1. A method for determining the direction of incidence of an incoming audio signal from an acoustic source to a directional microphone system, having at least two microphones, comprising the steps of:

detecting said incoming audio signal with said at least two microphones and, in each of said at least two microphones, producing an output microphone signal therefrom;

generating at least two phase-shifted directional microphone signals by phase shifting at least one output microphone signal relative to another output microphone signal and combining the phase-shifted output microphone signals with respective weightings, the respective weightings defining a direction-dependent sensitivity distribution, having a minimum in one direction, for the respective directional microphone signals;

assessing each of said directional microphone signals with respect to a quantity that indicates an influence, on the respective directional microphone signal, by the associated direction-dependent sensitivity distribution; and

comparing the respective quantities of the respective directional microphone signals with each other to identify a quantity having an extreme value, and determining the direction of incidence of said incoming audio signal as being the direction at which the minimum of the direction-dependent sensitivity distribution for the directional microphone signal having said extreme value is located.

2. A method as claimed in claim 1 comprising employing energy in the respective directional microphone signals as said quantity, and determining the direction of the minimum of the direction-dependent sensitivity distribution having the least energy as being said direction of incidence.

3. A method as claimed in claim 1 comprising employing a reciprocal of energy of the respective directional microphone signals as said quantity, said reciprocal of said energy representing a probability that the direction of the minimum of the direction-dependent sensitivity distribution of the directional microphone signal associated with the reciprocal is said direction of incidence.

4. A method as claimed in claim 3 comprising combining the respective probabilities of the directional microphone signals to form a direction-resolved probability distribution, and determining the direction of incidence of said incoming audio signal from said probability distribution.

5. A method as claimed in claim 1 comprising setting the respective weightings to minimize the sensitivity of the directional microphone system for a signal source located in a selected direction with respect to the directional microphone system.

6. A method as claimed in claim 1 comprising selecting said weighting to embody an effect of an acoustic environment in which said directional microphone system is being used.

7. A method as claimed in claim 6 comprising determining the respective weightings by measuring the sensitivity of the directional microphone system at a head or a head simulation.

8. A method as claimed in claim 1 wherein each of said microphone signals has an amplitude and a phase, and comprising employing a weighting having an amplitude factor and a phase factor for correcting at least one of the amplitude or the phase of at least one of said microphone signals.

9. A method as claimed in claim 1 comprising storing said weighting as a frequency-dependent characteristic.

10. A method as claimed in claim 1 comprising reading the respective weightings from a memory.

11. A method as claimed in claim 1 comprising generating said directional microphone signals substantially simultaneously.

12. A method as claimed in claim 1 comprising varying the respective weightings for two or more of said directional microphone signals to successively produce respective directional microphone signals having direction-dependent sensitivity distributions.

13. A method as claimed in claim 1 wherein each of said microphone signals has a frequency range, and comprising subdividing each frequency range into a plurality of frequency bands, each having a microphone signal component therein, and using said microphone signal components as said microphone signals.

14. A method as claimed in claim 13 comprising assessing the respective quantities of the respective directional microphone signals in at least two of said frequency bands.

15. A method as claimed in claim 1 comprising weighting the respective microphone signals from the microphones in said directional microphone system in pairs to produce said directional microphone signal.

16. A method as claimed in claim 1 wherein said incoming audio signal is a first incoming audio signal from a first source, and comprising detecting a second incoming audio signal from a second signal source with said microphones in said directional microphone system, and determining the direction of incidence of said second incoming audio signal from said quantity.

17. A method as claimed in claim 16 comprising assessing said quantities for said first and second incoming audio signals in a same frequency band by correlation.

18. A method as claimed in claim 17 comprising assessing said first and second incoming audio signals by correlation according to an echo relationship.

19. A method as claimed in claim 16 comprising assessing said quantities for said first and second incoming audio signals in respectively different frequency bands by correlation.

20. A method as claimed in claim 19 comprising assessing said first and second incoming audio signals by correlation according to an echo relationship.

21 A method as claimed in claim 1 comprising experimentally determining the direction of the minimum of each direction-dependent sensitivity distribution using an experimental signal source with said directional microphone system.

22. A method as claimed in claim 1 comprising determining the direction of the minimum of the direction-dependent sensitivity distribution by calculation with measured transfer functions.

23. An apparatus for determining a direction of incidence of an incoming audio signal comprising:

a directional microphone system having at least two microphones for detecting said incoming audio signal, each of said at least two microphones generating a microphone signal therefrom;

a phase-shifter that phase-shifts at least one microphone signal of said system relative to another microphone signal of said system;

weighting units for respectively weighting said phase-shifted microphone signals for producing at least two directional microphone signals, the respective weightings defining a direction-dependent sensitivity distribution for each of said directional microphone signals;

an assessment unit for assessing the respective directional microphone signals with respect to a quantity representing an influence of the direction-dependent sensitivity distribution on the directional microphone signal; and

a determination unit that identifies one of said directional microphone signals having an extreme value of said quantity compared to the other directional microphone signals, and for determining the direction of incidence of said incoming audio signal as being a direction in which a minimum of the direction-dependent sensitivity distribution of said one of said directional microphone signals is located.

24. An apparatus as claimed in claim 23 comprising, for each of said microphones, a filter bank connected thereto for subdividing the microphone signal from the microphone signal connected thereto into a plurality of frequency bands each frequency band having an output at which a signal component of the

microphone signal in that frequency band is present, with respective outputs of the respective filter banks in the same frequency band being connected in pairs to said weighting unit, said weighting unit comprising at least one of an amplitude unit for varying an amplitude of the signal component and a phase unit for shifting the phase of the signal component.

25. An apparatus as claimed in claim 24 wherein said weighting unit comprises both said amplitude units and said phase units, and wherein said amplitude units and said phase unit operate jointly on each signal component.

26. An apparatus as claimed in claim 24 wherein said assessment unit comprises a plurality of assessment subunits respectively operating in different ones of said frequency bands for assessing said quantity in the different frequency bands, and an analysis unit connected to said assessment subunits for generating, from the assessment of the quantities in the respectively different frequency bands, an acoustic environment analysis result.

27. An apparatus as claimed in claim 26 wherein said analysis result generates said acoustic environment analysis result by a correlation analysis of a time response in the different frequency bands.

EVIDENCE APPENDIX

Exhibit A: Figures 1, 2, 3 and 4 as originally filed on March 25, 2004

Exhibit B: United States Patent No. 6,069,961 (Nakazawa), cited in the Final Rejection dated July 17, 2008

Exhibit C: United States Patent No. 6,584,203 (Elko et al.), cited in the Final Rejection dated July 17, 2008

RELATED PROCEEDINGS APPENDIX

None.

CH1\6102303.1

FIG 1

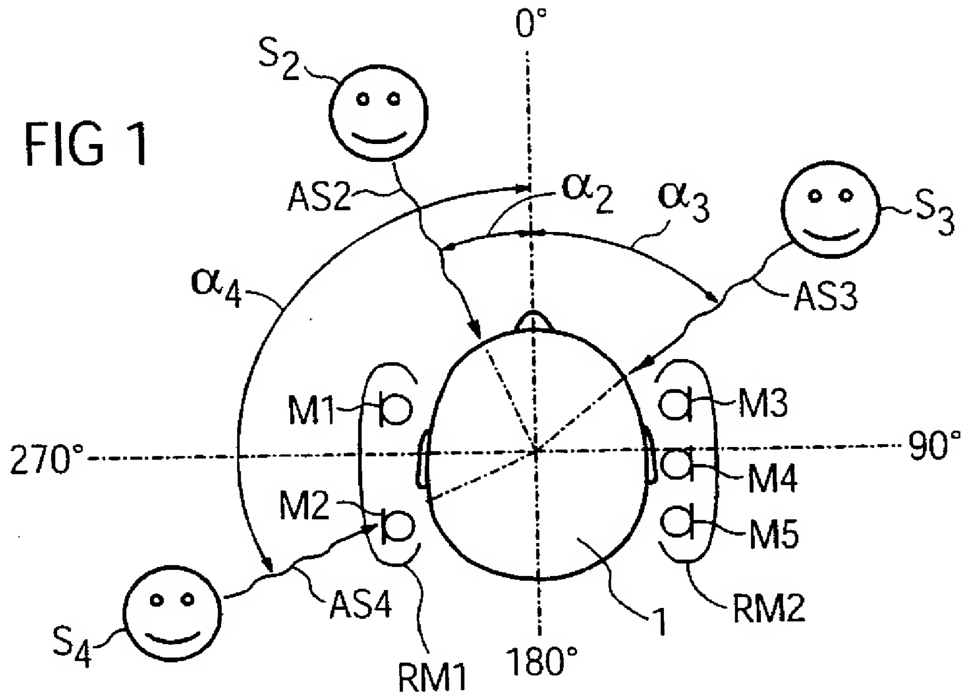


FIG 2

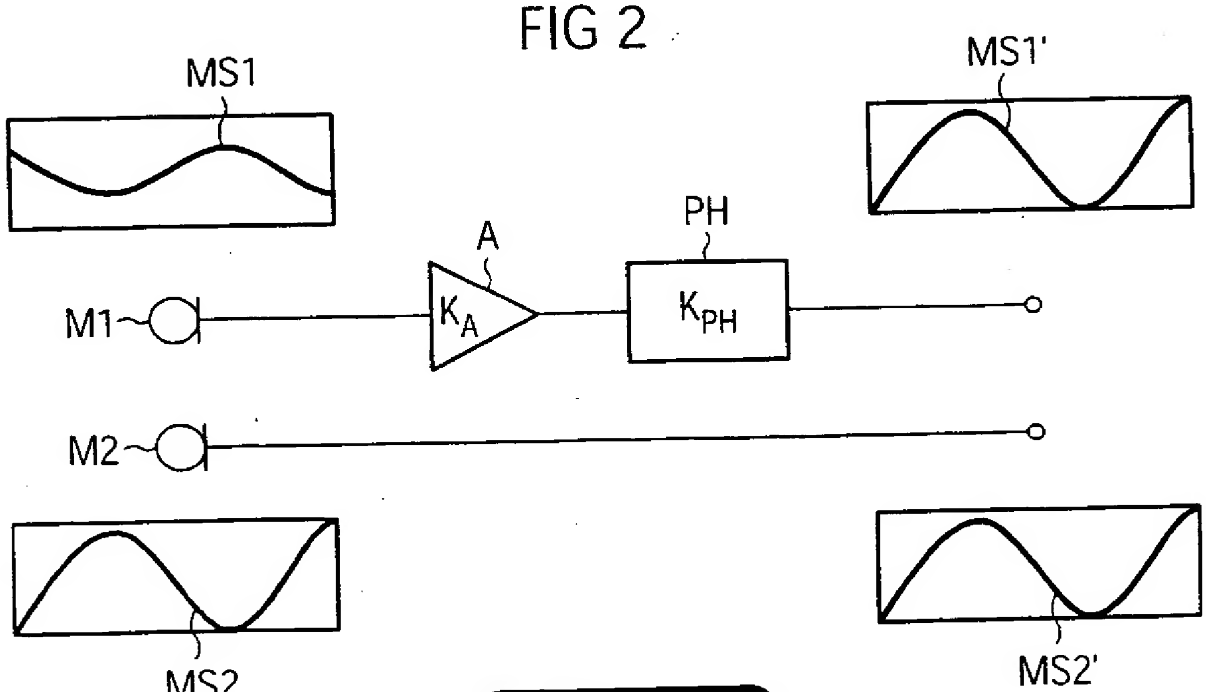


EXHIBIT
A
ALL-STATE LEGAL®

FIG 3

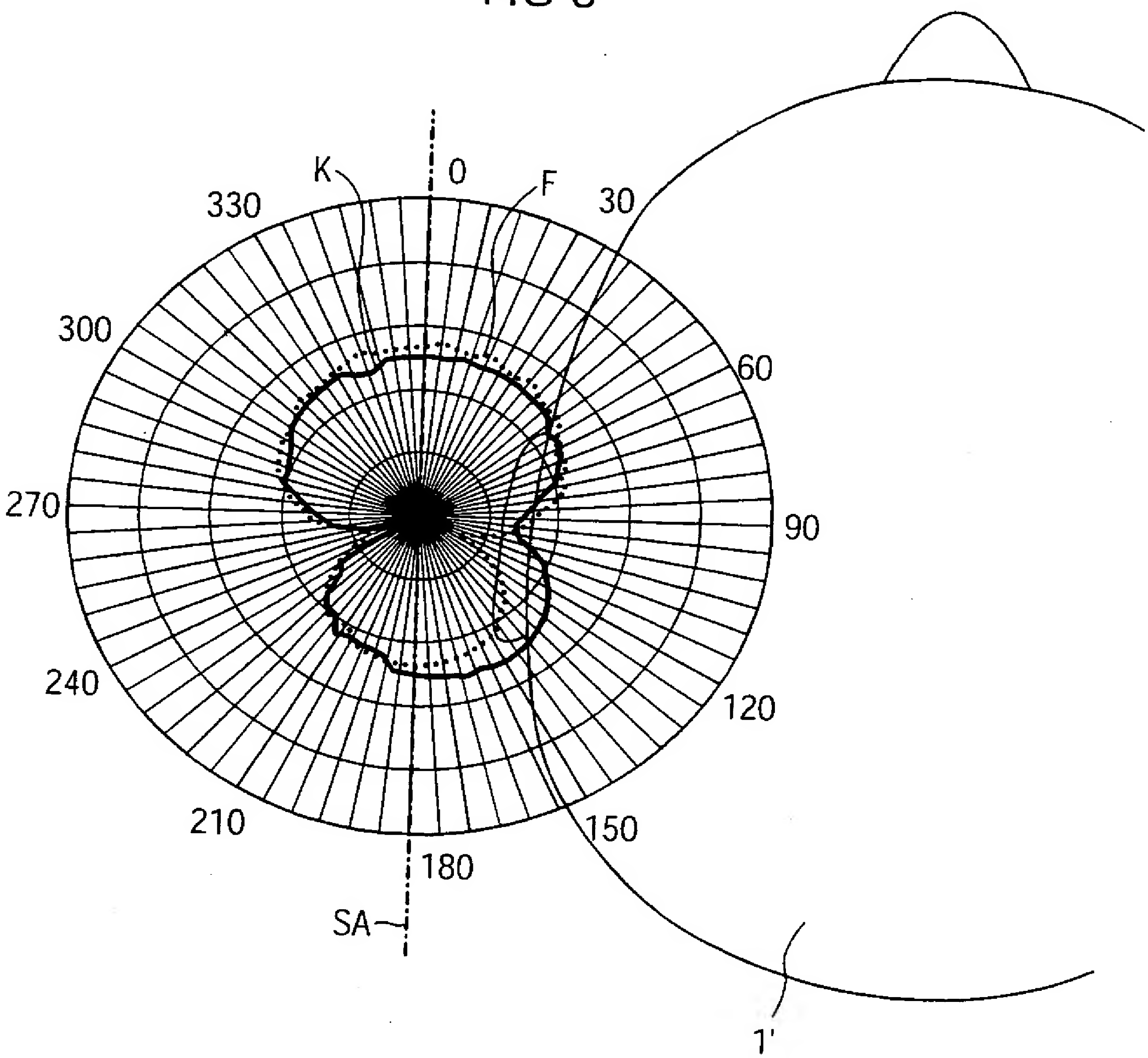
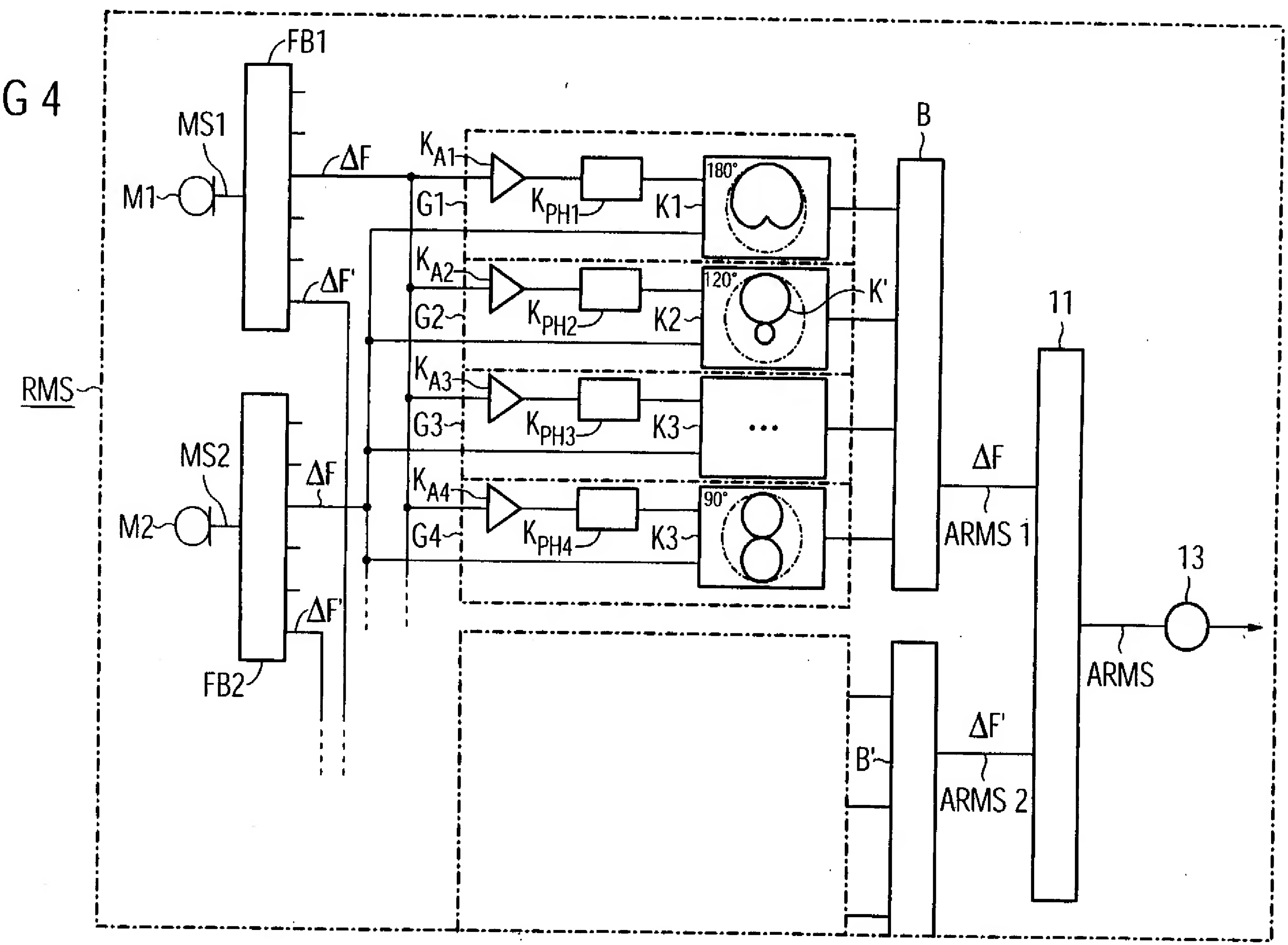


FIG 4



RMS
FB1
MS1
M1
ΔF
ΔF'
MS2
M2
FB2
K A1
G1
K PH1
K1
180°
K A2
G2
K PH2
K2
120°
K'
K A3
G3
K PH3
K3
...
K A4
G4
K PH4
K3
90°
B
B'
ΔF
ARMS 1
ΔF'
ARMS 2
11
ARMS
13

US006069961A

United States Patent [19]

Nakazawa

[11] **Patent Number: 6,069,961**

[45] **Date of Patent: May 30, 2000**

[54] **MICROPHONE SYSTEM**

[75] Inventor: **Fumihiko Nakazawa**, Kawasaki, Japan

[73] Assignee: **Fujitsu Limited**, Kawasaki, Japan

[21] Appl. No.: **08/965,191**

[22] Filed: **Nov. 6, 1997**

[30] **Foreign Application Priority Data**

Nov. 27, 1996 [JP] Japan 8-316567
Jun. 12, 1997 [JP] Japan 9-155246

[51] **Int. Cl.**[7] **H04R 3/00**

[52] **U.S. Cl.** **381/92**; 381/122

[58] **Field of Search** 381/92, 122; 367/124, 367/126

[56] **References Cited**

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4-72525	3/1992	Japan .
5-207117	8/1993	Japan .
7-336790	12/1995	Japan .

Primary Examiner—Vivian Chang
Assistant Examiner—Debra A. Lemm
Attorney, Agent, or Firm—Armstrong, Westerman, Hattori, McLeland & Naughton

[57] **ABSTRACT**

Provided is a microphone system capable of detecting a direction of a sound source and extracting an object sound with a high S/N ratio. A microphone system comprises a plurality of microphone pairs 1 to 7, each pair having two microphones arranged apart from each other at a predetermined space at a crossing angle of 60 degrees or less, a plurality of subtraction circuits **11***a* for calculating a difference signal of outputs of each microphone pair, a plurality of addition circuits **11***e* for calculating a sum signal of outputs of each microphone pair, circuit sections **11***c*, **11***d* and **20** for detecting, as sound source direction information, a minimum value output from each output of the subtraction circuits **11***a*, and a switch **11***f* for selecting a sum signal of the microphone pairs **1** to **7** corresponding to the minimum value output and for outputting the selected sum signal as sound information.

8 Claims, 10 Drawing Sheets

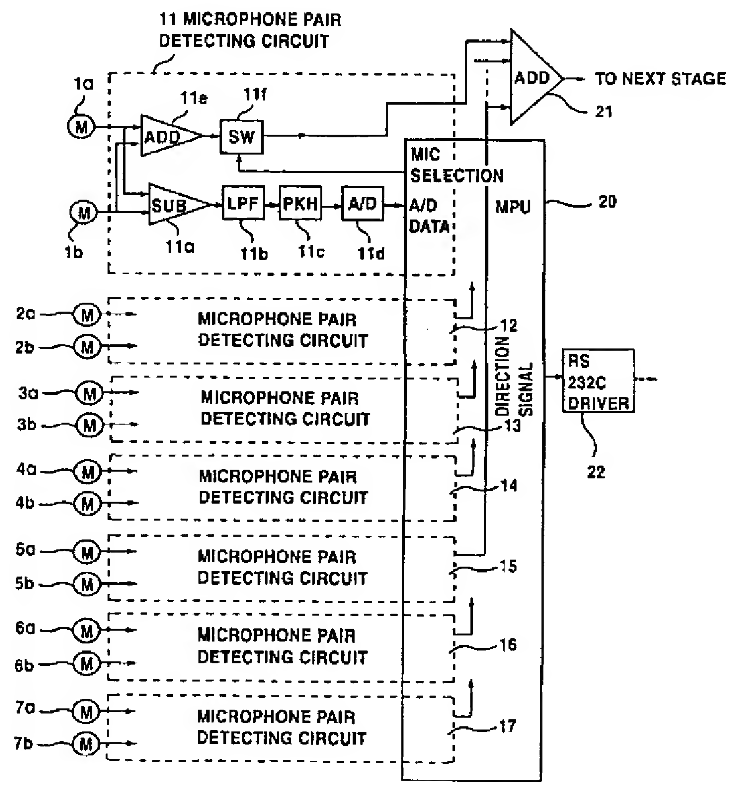

EXHIBIT B
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FIG.1A

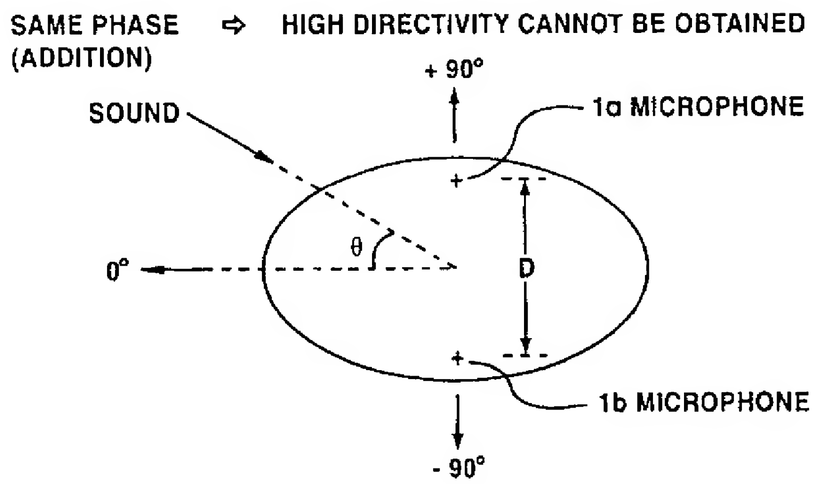

SAME PHASE (ADDITION)
⇨
HIGH DIRECTIVITY CANNOT BE OBTAINED
+ 90°
SOUND
1a MICROPHONE
θ
0°
D
1b MICROPHONE
- 90°

FIG.1B

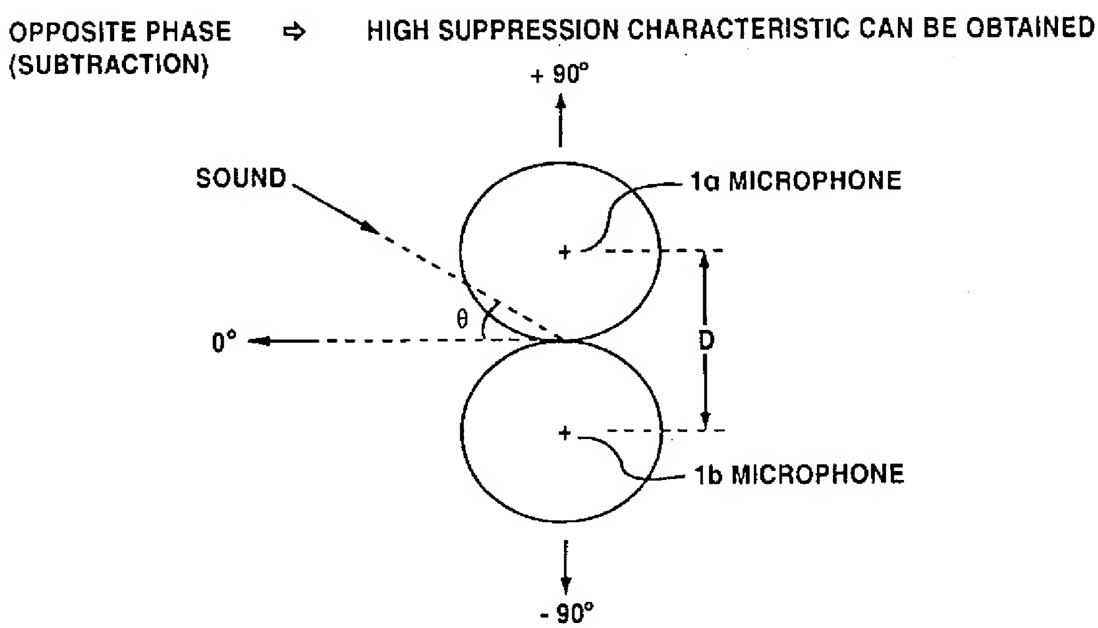

OPPOSITE PHASE (SUBTRACTION)
⇨
HIGH SUPPRESSION CHARACTERISTIC CAN BE OBTAINED
+ 90°
SOUND
1a MICROPHONE
θ
0°
D
1b MICROPHONE
- 90°

FIG.2A

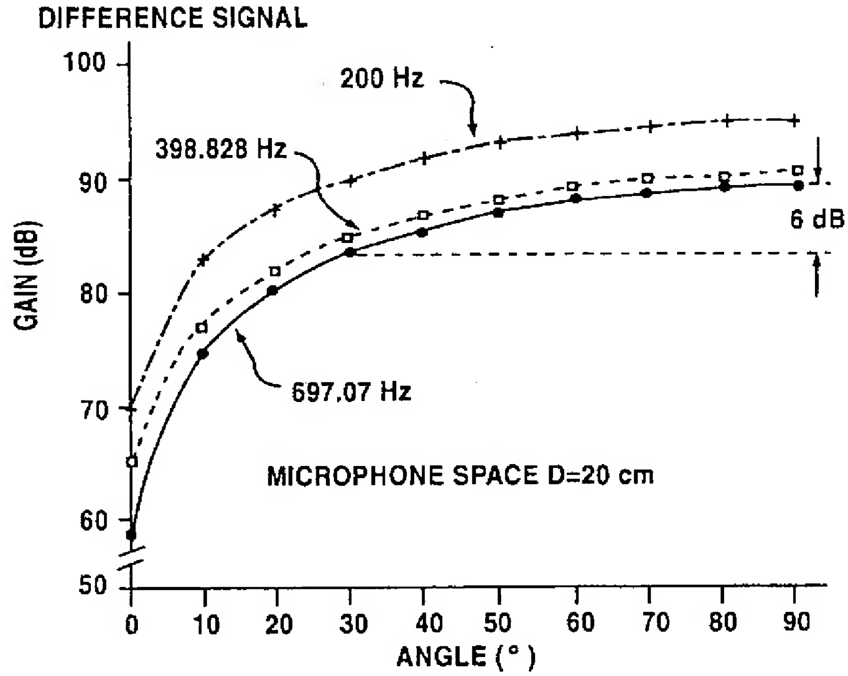

DIFFERENCE SIGNAL
GAIN (dB)
100
90
80
70
60
50
200 Hz
398.828 Hz
697.07 Hz
6 dB
MICROPHONE SPACE D=20 cm
0
10
20
30
40
50
60
70
80
90
ANGLE (°)

FIG.2B

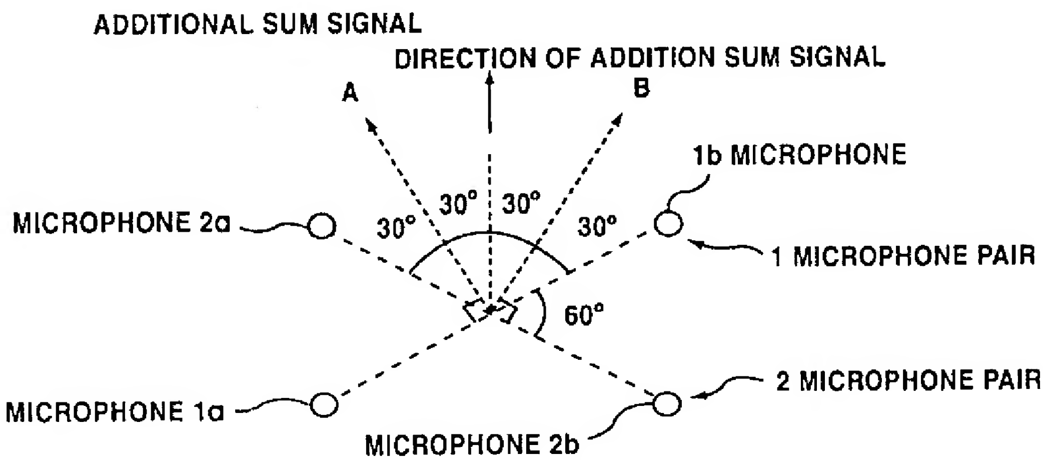

ADDITIONAL SUM SIGNAL
DIRECTION OF ADDITION SUM SIGNAL
A
B
1b MICROPHONE
MICROPHONE 2a
30°
30°
30°
30°
1 MICROPHONE PAIR
60°
MICROPHONE 1a
2 MICROPHONE PAIR
MICROPHONE 2b

FIG.3A

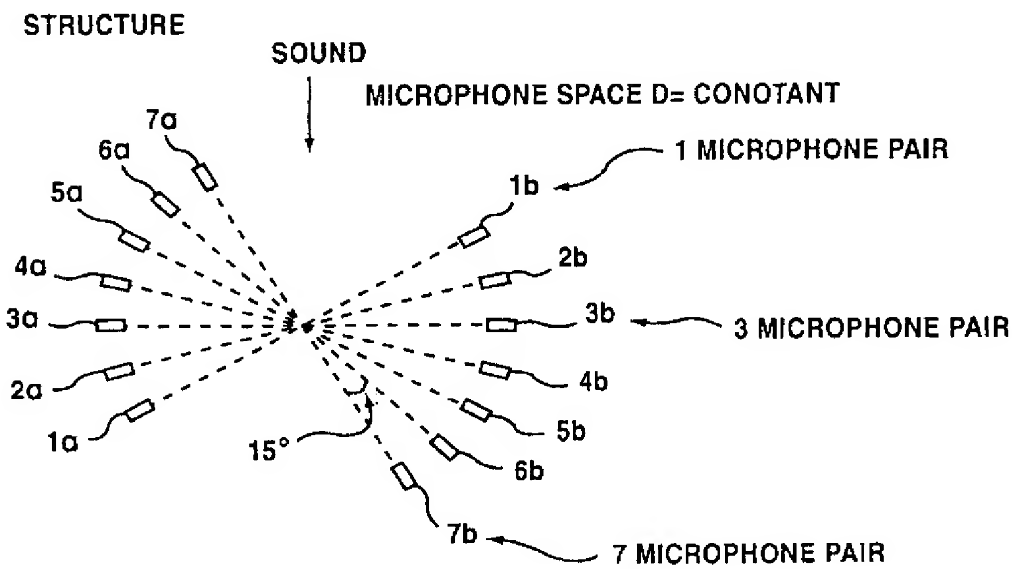

STRUCTURE
SOUND
MICROPHONE SPACE D= CONOTANT
1 MICROPHONE PAIR
3 MICROPHONE PAIR
7 MICROPHONE PAIR
7a
6a
5a
4a
3a
2a
1a
1b
2b
3b
4b
5b
6b
7b
15°

FIG.3B

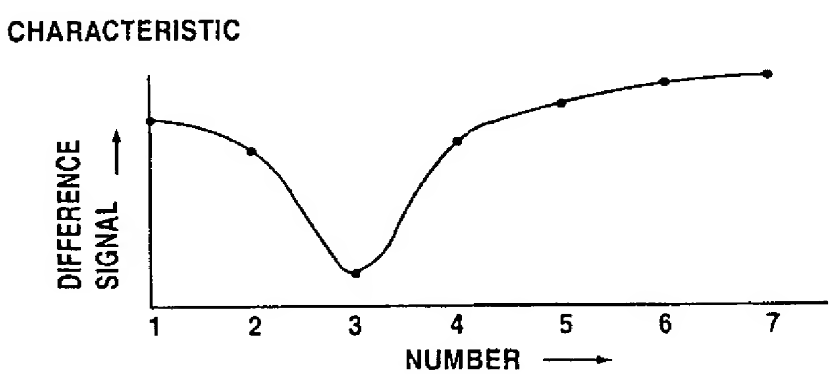

CHARACTERISTIC
DIFFERENCE SIGNAL
1
2
3
4
5
6
7
NUMBER

FIG.4

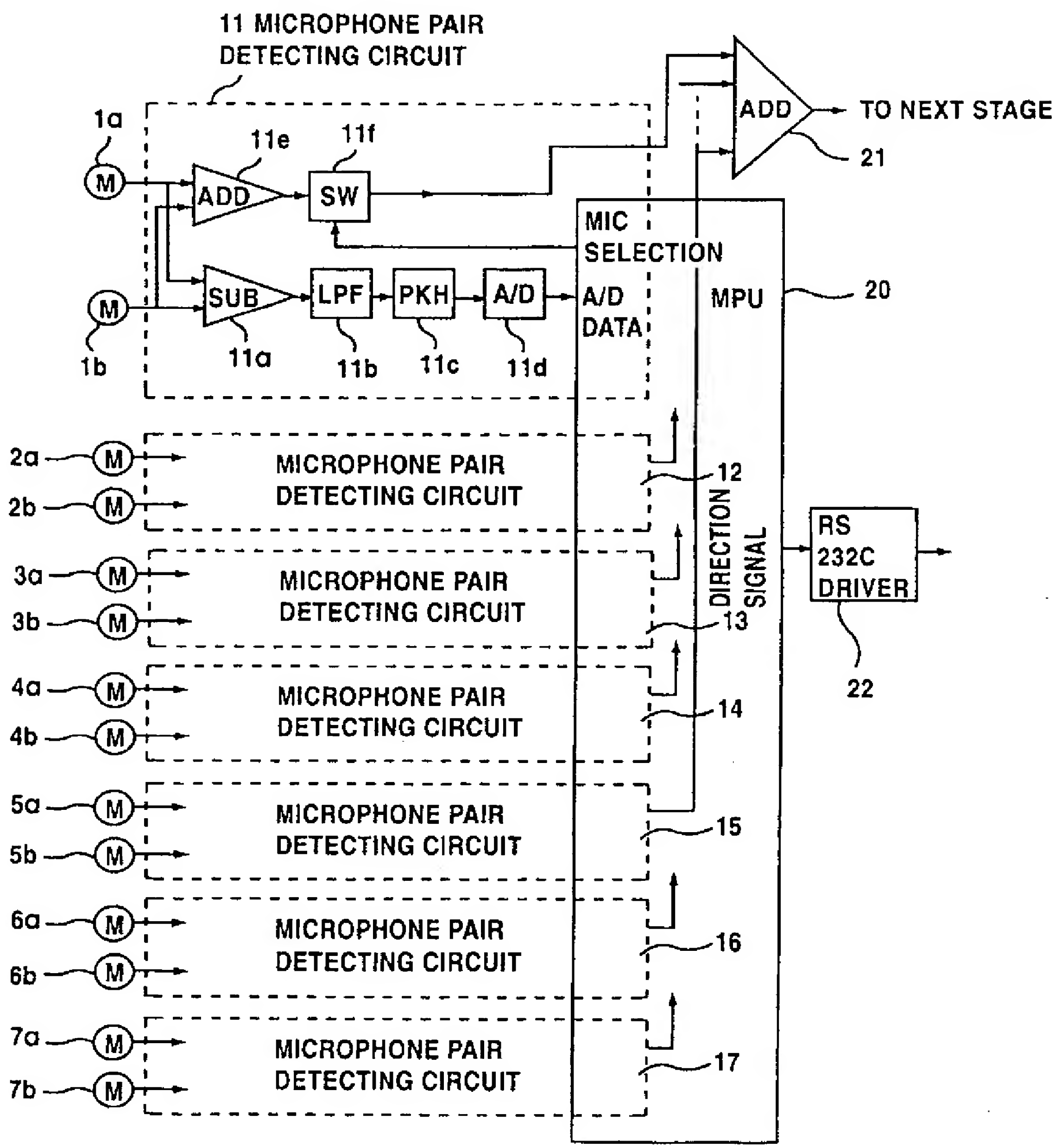

11 MICROPHONE PAIR DETECTING CIRCUIT
1a
1b
M
11e
ADD
11f
SW
SUB
LPF
PKH
A/D
11a
11b
11c
11d
MIC SELECTION
A/D DATA
MPU
20
ADD
TO NEXT STAGE
21
2a
2b
MICROPHONE PAIR DETECTING CIRCUIT
12
3a
3b
MICROPHONE PAIR DETECTING CIRCUIT
13
4a
4b
MICROPHONE PAIR DETECTING CIRCUIT
14
5a
5b
MICROPHONE PAIR DETECTING CIRCUIT
15
6a
6b
MICROPHONE PAIR DETECTING CIRCUIT
16
7a
7b
MICROPHONE PAIR DETECTING CIRCUIT
17
DIRECTION SIGNAL
RS 232C DRIVER
22

FIG.5A

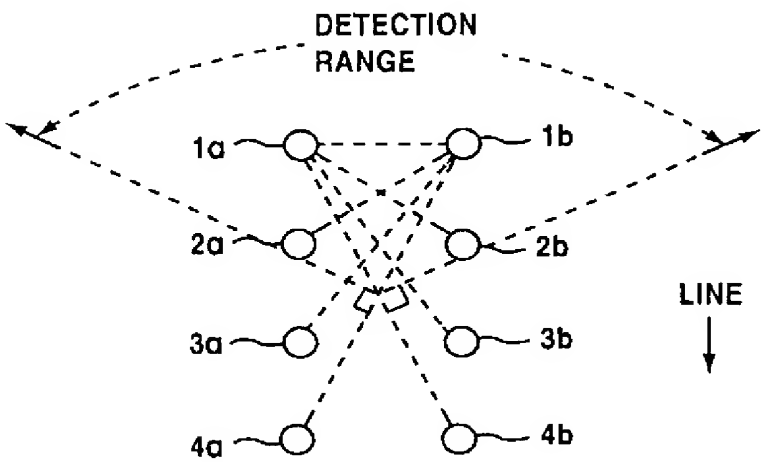

DETECTION
RANGE
1a
1b
2a
2b
3a
3b
4a
4b
LINE

FIG.5B

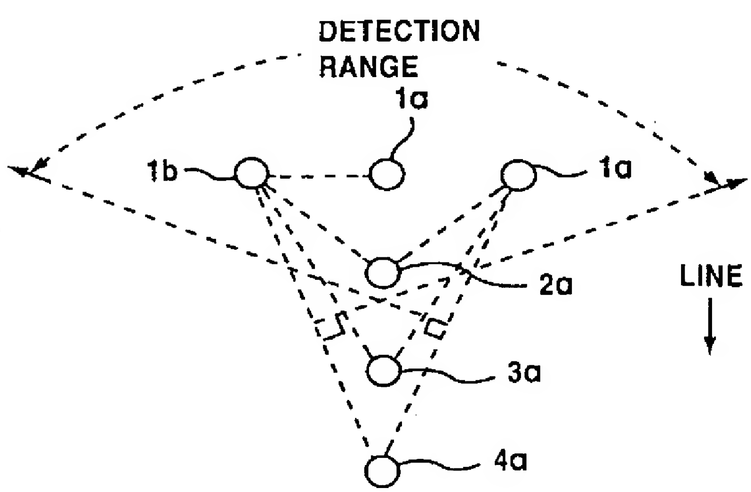

DETECTION
RANGE
1a
1b
1a
2a
3a
4a
LINE

FIG.6

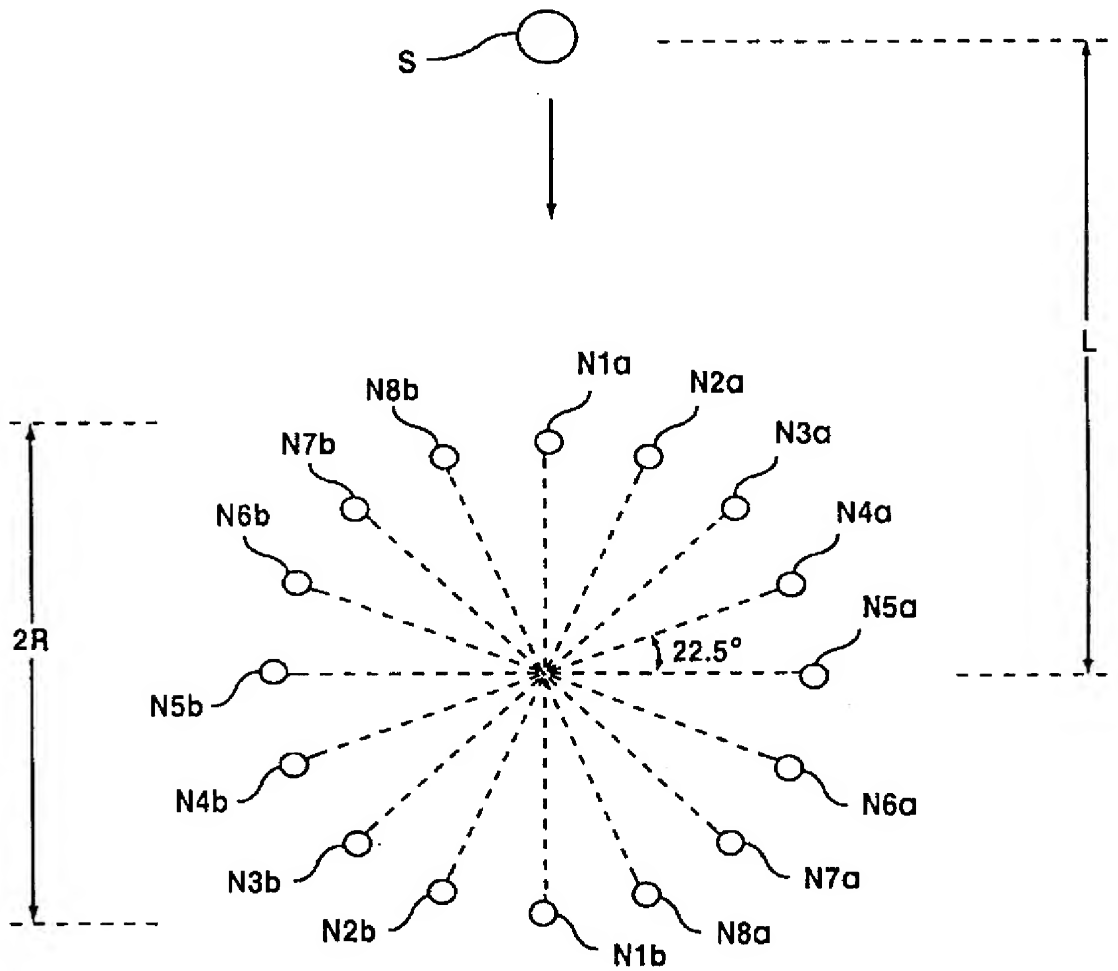

S
L
N1a
N2a
N3a
N4a
N5a
N6a
N7a
N8a
N1b
N2b
N3b
N4b
N5b
N6b
N7b
N8b
22.5°
2R

FIG.7

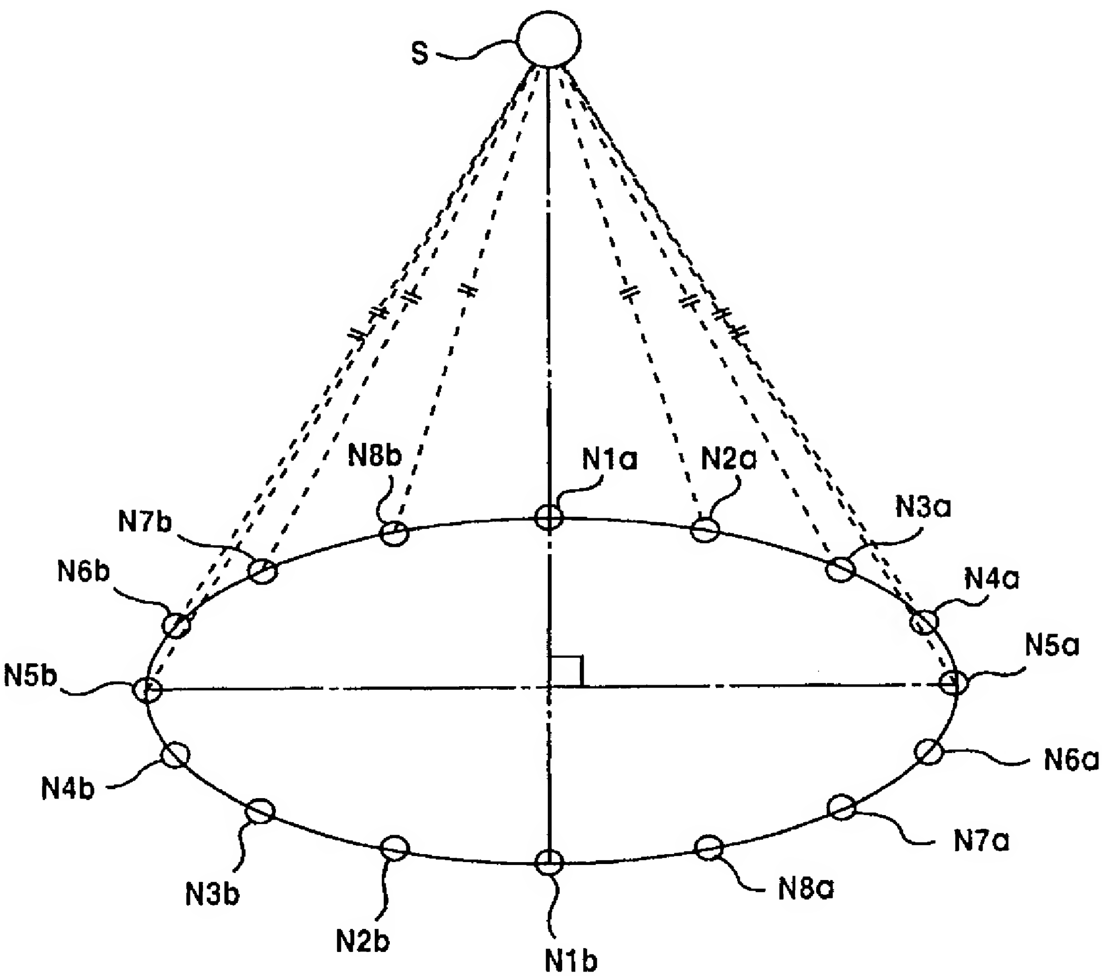


FIG.8

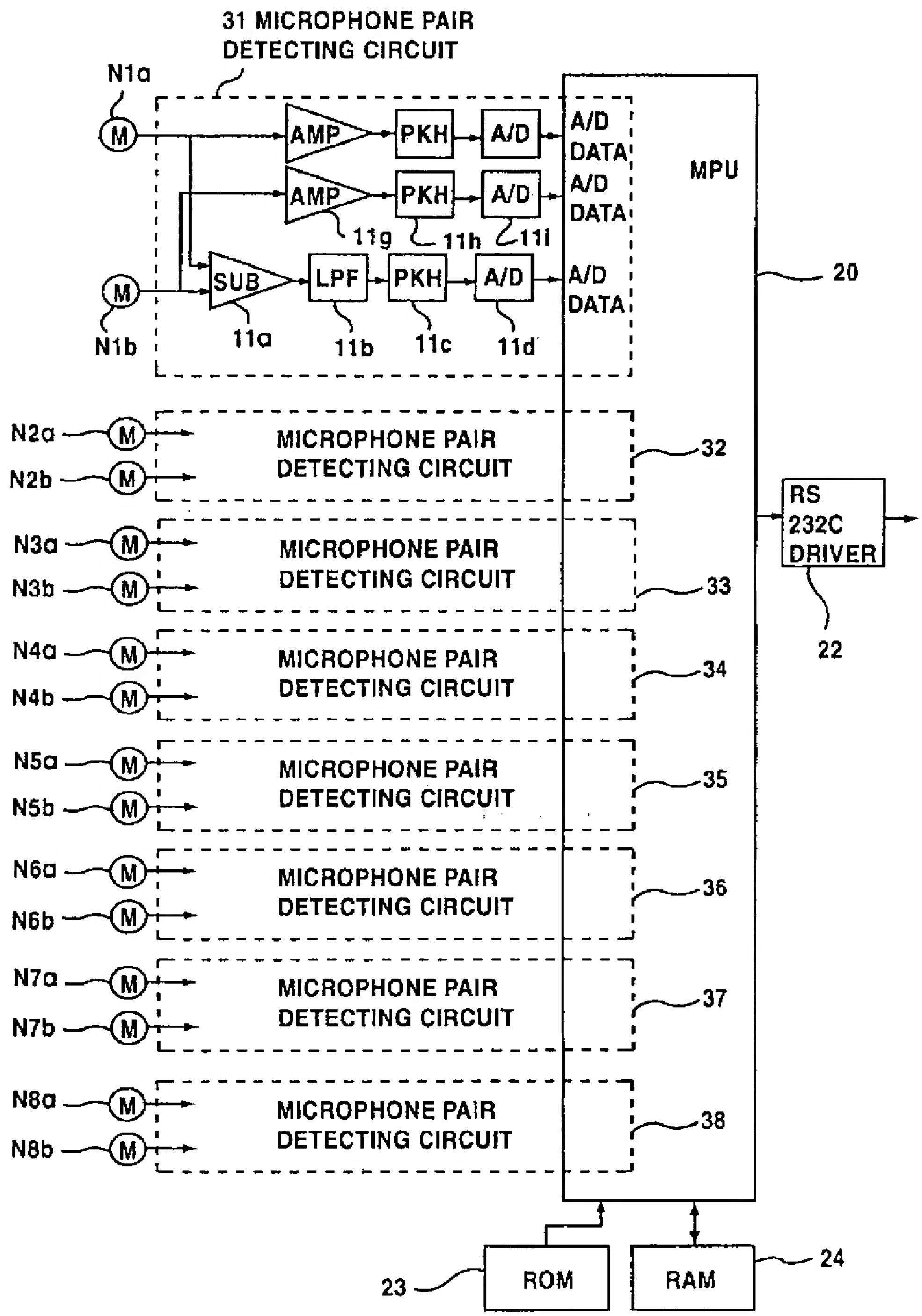

31 MICROPHONE PAIR DETECTING CIRCUIT
N1a
M
AMP
PKH
A/D
A/D DATA
AMP
PKH
A/D
A/D DATA
11g
11h
11i
SUB
LPF
PKH
A/D
A/D DATA
M
N1b
11a
11b
11c
11d
MPU
20
N2a
N2b
MICROPHONE PAIR DETECTING CIRCUIT
32
RS 232C DRIVER
N3a
N3b
MICROPHONE PAIR DETECTING CIRCUIT
33
22
N4a
N4b
MICROPHONE PAIR DETECTING CIRCUIT
34
N5a
N5b
MICROPHONE PAIR DETECTING CIRCUIT
35
N6a
N6b
MICROPHONE PAIR DETECTING CIRCUIT
36
N7a
N7b
MICROPHONE PAIR DETECTING CIRCUIT
37
N8a
N8b
MICROPHONE PAIR DETECTING CIRCUIT
38
23
ROM
RAM
24

FIG.9

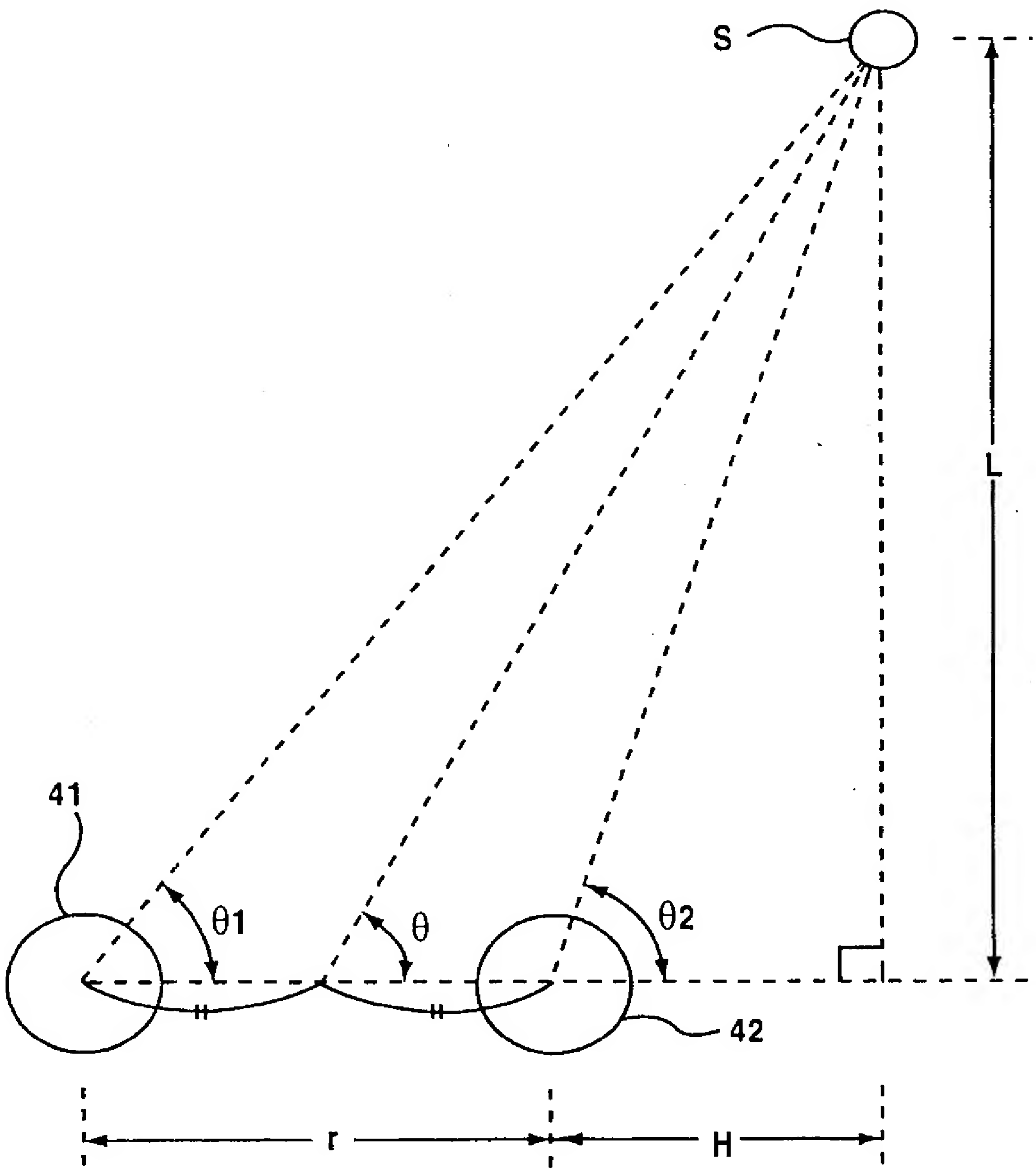
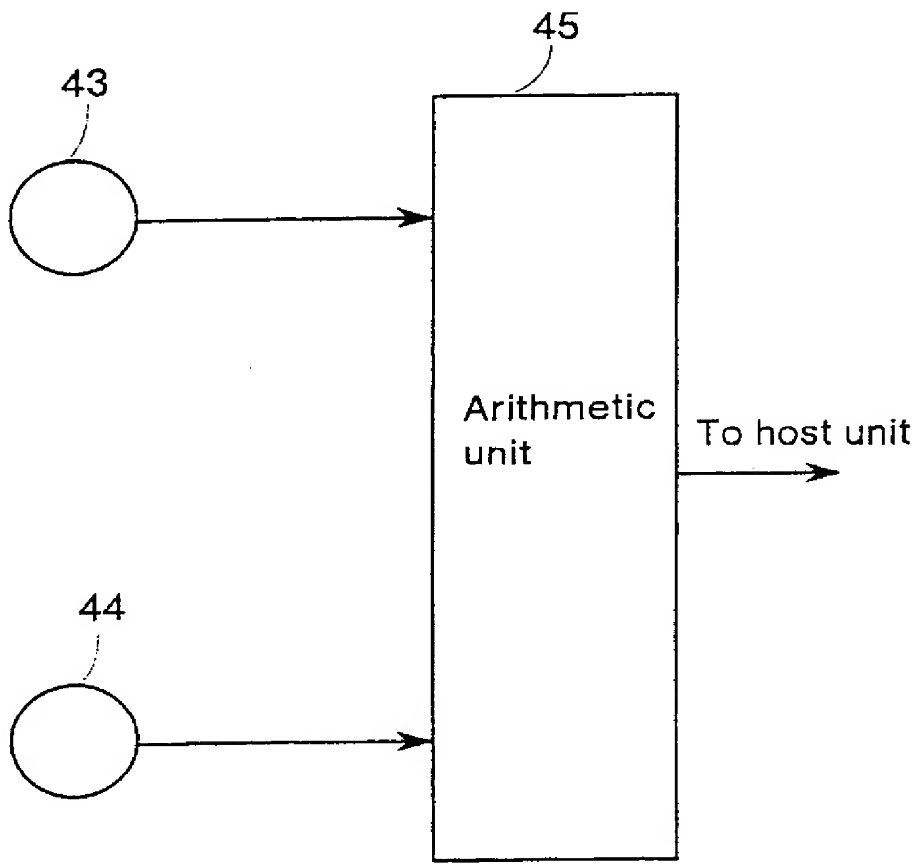


FIG.10



MICROPHONE SYSTEM

CROSS-REFERENCES TO RELATED APPLICATIONS

This application is related to Japanese Patent Application Nos. Hei 8(1996)-316567 filed on Nov. 27, 1996 and Hei 9(1997)-155246 filed on Jun. 12, 1997 whose priorities are claimed under 35 USC Section 119, the disclosures of which are incorporated herein by reference in their entirety.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to a microphone system to be used for an interface technique of a talker and a personal computer, and more particularly to a microphone system in which a position of a sounder (talker) is input as input data to a personal computer and a signal-to-noise (S/N) ratio of a detecting signal of pronunciation is enhanced so that a sound processing (for example, voice recognition) in a next stage can be improved.

2. Description of the Related Art

Examples of the prior art in which a direction of a sound source is detected by using a plurality of microphones include Japanese Unexamined Patent Publication Nos. HEI 4(1992)-72525, HEI 5(1993)-207117 and HEI 7(1995)-336790.

According to the Japanese Unexamined Patent Publication No. HEI 4(19-72525, a spherical received sound detecting section having an integral structure in which six nondirectional microphones are arranged at the space of 90 degrees seen from a center of a sphere on surface thereof is used to detect a received sound pressure level of each microphone and a difference signal of opposite microphones, and to calculate a direction of a sound source based on the received sound pressure level and the difference signal so that the direction of the sound source present in a three-dimensional space can easily be obtained with high precision.

Since only the direction of the sound source is detected by using the six nondirectional microphones, said Publication does not disclose that a S/N ratio is improved to extract an object sound.

According to the Japanese Unexamined Patent Publication No. HEI 5(1993)-207117, a talker's voice, that is, a driver's voice to be input to a mobile phone is received by using at least three microphones for detecting his position, a time difference between voice signals is detected, a position of the talker is detected based on the time difference, and a directional microphone is provided in a direction of the talker so that the influence of noises is reduced to enhance recognition of the talker's voice. In said method, three or more superdirectional microphones are used for extracting an object sound and are provided in a direction of the talker. In general, the superdirectional microphone has a total length of 50 cm or more in order to obtain a high directivity. Furthermore, there is no description on an improvement in the S/N ratio to extract the object sound.

According to the Japanese Unexamined Patent Publication No. HEI 7(1995)-336790, a plurality of microphones are provided to select a microphone in which an output has a maximum value or a generation timing is the earliest so that manual operation of the microphone, an interference of a sound signal and manual operation of mixing can automatically be performed and improved. In said method, a single microphone output is used for extracting a direction of a sound source and an object sound. There is also no description on an improvement in the S/N ratio to extract the object sound.

Therefore, the microphone systems according to the prior arts have problems that a structure is not always small and simple and an object sound in a direction of a sound source cannot be extracted with a high S/N ratio.

SUMMARY OF THE INVENTION

The present invention aims to provide a microphone system having a small and simple structure and capable of detecting a direction of a sound source and extracting an object sound with a high S/N ratio.

In order to attain the above-mentioned object, as shown in FIGS. 3A, 3B and 4, the present invention provides a microphone system comprising: a plurality of microphone pairs, each pair having two microphones arranged apart from each other at a predetermined space at a crossing angle of 60 degrees or less; a plurality of first calculating means for calculating a difference signal of outputs of each microphone pair; a plurality of second calculating means for calculating a sum signal of outputs of each microphone pair; means for detecting, as sound source direction information, a minimum value output from each output of the first calculating means; and means for selecting a sum signal of the microphone pair corresponding to the minimum value output and outputting the selected sum signal as sound information.

BRIEF DESCRIPTION OF THE DRAWINGS

FIGS. 1A and 1B are diagrams showing direction characteristics of a microphone pair having two non-directional microphones;

FIGS. 2A and 2B are charts showing characteristics of a difference signal and a sum signal of the microphone pair;

FIGS. 3A and 3B are charts showing a structure and a characteristic of a microphone pair according to a first embodiment of the present invention;

FIG. 4 is a diagram showing a structure of a microphone pair detecting circuit according to an embodiment of the present invention;

FIGS. 5A and 5B are diagrams showing structures of a microphone pair according to a second embodiment of the present invention;

FIG. 6 is a diagram showing a structure of a microphone pair according to a third embodiment of the present invention;

FIG. 7 is a diagram showing sensitivity adjustment of the microphones according to the present invention;

FIG. 8 is a block diagram showing a signal processing circuit according to the third embodiment of the present invention;

FIG. 9 is a diagram showing a structure of a microphone pair according to a fourth embodiment of the present invention; and

FIG. 10 is a diagram showing a circuit structure using two microphone arrays according to the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to FIGS. 3 and 4, the microphone system of the present invention comprises a plurality of microphone pairs 1 to 7, each pair having two microphones arranged apart from each other at a predetermined space at a crossing angle

of 60 degrees or less, a plurality of subtraction circuits **11***a* for calculating a difference signal of outputs of each microphone pair, a plurality of addition circuits **11***e* for calculating a sum signal of outputs of each microphone pair, circuits **11***c*, **11***d* and **20** for detecting, as sound source direction information, a minimum value output from each output of the subtraction circuits **11***a*, and a switch **11***f* for selecting a sum signal of the microphone pair corresponding to the minimum value output and outputting the selected sum signal as sound information.

In the present invention, it is desirable that the switch **11***f* should also select and output a sum signal corresponding to a second smallest difference signal in addition to a sum signal corresponding to a minimum value of a difference signal as shown in FIGS. **3**A and **3**B and FIG. **4**.

As shown in FIGS. **3**A and **3**B, it is preferable that the plurality of microphone pairs **1** to **7** should be arranged on a concentric circle.

As shown in FIG. **4**, it is desirable that the microphone system should further comprises a low-pass filter **11***b* for filtering a difference signal output from a subtraction circuit **11***a* at a cut-off frequency F represented by V/2D=F between a sound velocity V and a space D.

Furthermore, the present invention provides a microphone system comprising a microphone array in which microphone sets are arranged at a crossing angle of 45 degrees or less, each microphone set including a first microphone pair having two microphones arranged apart from each other at a predetermined space and a second microphone pair orthogonal to the first microphone pair, a plurality of first calculating means for calculating a difference signal of outputs of each microphone pair, means for detecting, as a direction of a sound source, a minimum value output from each output of the first calculating means, second calculating means for calculating a ratio of output voltages of a microphone pair orthogonal to a microphone pair corresponding to the minimum value output, and means for calculating a distance from the microphone array to a sound source based on the ratio of the output voltages which is calculated by the second calculating means.

Moreover, the present invention provides a microphone system comprising a plurality of sound source direction discriminators including a plurality of microphone pairs, each pair having two microphones arranged apart from each other at a predetermined space at a crossing angle of 60 degrees or less, a plurality of calculating means for calculating a difference signal of outputs of each microphone pair, and means for detecting, as a direction of a sound source, a minimum value output from each output of the calculating means, wherein a crossing point of the directions of the sound source which are obtained by the sound source direction discriminators is calculated, thereby detecting a position of the sound source.

In the above-mentioned microphone system, it is desirable that a non-directional microphone should be used for the microphone. In this case, it is preferable that a sound source should be provided on a central axis of the microphone array to adjust a sensitivity of each microphone of the microphone array equally.

Preferred embodiments of the present invention will be described below with reference to FIGS. **1**A and **1**B to FIGS. **5**A and **5**B.

First of all, basic matters related to the present invention will be described below with reference to FIGS. **1**A and **1**B and FIGS. **2**A and **2**B.

FIGS. **1**A and **1**B are diagrams showing direction characteristics of a microphone pair having two non-directional microphones. FIG. **1**A shows a case in which the two non-directional microphones (hereinafter referred to as microphones) are used in the same phase (addition), and FIG. **1**B shows a case in which the microphones are used in opposite phases (subtraction).

The two microphones have the same characteristics (a directivity of a sensitivity and a frequency characteristic), and are provided with a space D.

As shown in FIGS. **1**A and **1**B, it is assumed that two microphones **1***a* and **1***b* are arranged coaxially to form a microphone pair. If the microphones **1***a* and **1***b* have the same phase (addition) as shown in FIG. **1**A, a direction characteristic of a sum signal of the microphones **1***a* and **1***b* in an arrival direction θ of a sound is elliptical.

Accordingly, a great difference is not made between a gain of a direction of 0 degree (a direction perpendicular to directions of arrangement of the microphones **1***a* and **1***b*) and a gain of a direction of 90 degrees (the directions of arrangement of the microphones **1***a* and **1***b*). Consequently, a high direction characteristic cannot be obtained.

However, if the microphones **1***a* and **1***b* have the opposite phases (subtraction) as shown in FIG. **1**B, the gain in the direction of 0 degree is suppressed so that a direction characteristic of a difference signal of the microphones **1***a* and **1***b* becomes almost "8" shaped.

Consequently, a great difference is made between the gains in the directions of 0 degree and 90 degrees. Thus, a high direction characteristic can be obtained.

According to the present invention, a direction of a sound source is detected by utilizing a high suppression directivity obtained in case of the opposite phases (subtraction).

Explanation will be given with reference to FIGS. **2**A and **2**B. FIGS. **2**A and **2**B show characteristics of difference and sum signals of the microphone pair.

FIG. **2**A shows the difference signal, that is, measured values of direction characteristics (a direction of a sound source and a gain) obtained by using two microphones in opposite phases.

A space D between the microphones is 20 cm, and a frequency of a sound is 200 Hz, 398.828 Hz and 697.07 Hz.

In FIG. **2**A, a measuring angle is 90 degrees in a direction of arrangement of the microphone pair. As shown, the gain is the smallest at an angle of 0 degree, and is increased as the angle becomes greater.

Within a range of 0 degree to 30 degrees, the gain is changed by at least 4 dB with an increase in an angle by 10 degrees. However, the gain is changed by about 2 dB or less with the increase in the angle by 10 degrees within a range of 30 degrees or more. In addition, the gain is changed by about 6 dB with the increase in the angle by 10 degrees within a range of 30 degrees to 90 degrees.

More specifically, if a sound arrives in a direction inclined by 30 degrees or more from a front of the microphone pair, the gain is less changed and outputs are rarely changed even if the angle is changed.

Accordingly, in the case where the difference signals of the two microphone pairs whose directions of a sound source are inclined by 30 degrees or more are compared with each other, it is impossible to accurately decide which microphone pair has an inclination closer to the direction of the sound source. In other words, the direction of the sound source cannot be detected accurately.

This indicates that it is necessary to set an angle of the microphone pair in the direction of the sound source to 30 degrees or less at worst. In other words, a crossing angle of

two microphone pairs should be set to 60 degrees or less in order to detect an accurate direction of the sound source.

Description will be given with reference to FIG. 2B. FIG. 2B is a chart showing an addition sum signal which is obtained by adding sum signals of the microphone pairs **1** and **2**.

It is assumed that the microphones **1**a and **1**b have the same phase and that the microphones **2**a and **2**b also have the same phase. Accordingly, the microphone pairs **1** and **2** have the same characteristics. A crossing angle of the microphone pairs **1** and **2** is set to 60 degrees.

As shown in FIG. 2B, a direction A of a sum signal of the microphone pair **1** is set to 30 degrees clockwise in a direction of arrangement of the microphone pair **2** which is about twice as much as an output of the microphone **1**a, for example.

Similarly, a direction B of a sum signal of the microphone pair **2** is set to 30 degrees counterclockwise in a direction of arrangement of the microphone pair **1** which is about twice as much as the output of the microphone **1**a, for example.

Accordingly, the directions of addition sum signals of the microphone pairs **1** and **2** are set to 30 degrees clockwise in a direction A, and to 30 degrees counterclockwise in a direction B, each of which is about four times as much as the output of the microphone **1**a, for example.

In general, n signals are synchronously added so that an amplitude becomes n times as much, while n random ambient noises are added so that the amplitude becomes square root times as large as n.

If the microphone pairs **1** and **2** are provided in the direction of the sound source, for example, sounds sent from the sound source are synchronously added. An output of the addition has an amplitude which is a multiple of the number of microphones to be added. The ambient noises are random. Therefore, the amplitude of the ambient noises becomes square root times as large as the number of the microphones to be added.

For example, in the case where a S/N ratio of an output of the microphone **1**a is compared with that of an output of the microphone pair **1** as described above, S is double at the maximum and N is 1.414 times as much at most. Therefore, the S/N ratio is improved by 3 dB at the maximum which is 1.414 times as much.

Accordingly, the microphone la is improved by 6 dB at the maximum.

Preferred embodiments of the present invention will be described below with reference to FIGS. 3A and 3B to FIGS. 5A and 5B.

FIGS. 3A and 3B show a structure and a characteristic of a microphone pair according to an embodiment of the present invention. FIG. 3A shows the structure and FIG. 3B shows the characteristic. FIG. 4 is a diagram showing a structure of a microphone pair detecting circuit according to a first embodiment of the present invention, which corresponds to seven microphone pairs **1** to **7** shown in FIG. 3A. FIGS. 5A and 5B shows structures of a microphone pair according to a second embodiment of the present invention.

[First Embodiment]

In FIG. 3A, seven microphone pairs **1** (microphones **1**a and **1**b) to **7** (microphones **7**a and **7**b) are provided, and a crossing angle of adjacent microphone pairs is set to 15 degrees, for example.

If a sound arrives in a direction of an arrow shown in FIG. 3A, respective levels of difference signals of the microphone pairs **1** to **7** are obtained as shown in a graph of FIG. 3B.

As shown in FIG. 3A, a third microphone pair **3** has a direction of arrangement which is almost perpendicular to a direction of arrival of the sound, and has a level of a difference signal which is the lowest as shown in FIG. 3B.

Furthermore, the level of the difference signal is increased in order of the microphone pairs **2**, **4**, **1**, **5**, . . . , **7**, for example.

More specifically, the levels of the difference signals of the microphone pairs **1** to **7** are compared with one another. By selecting the microphone pair **3** having the lowest level, the direction of arrival of the sound can be detected.

After the direction of arrival of the sound is detected, an object sound is then detected. By adding sum signals of the microphone pairs whose directions of arrival of the sound have been detected, the object sound is detected.

A subtraction output generated from the difference signal of the microphone pairs is filtered by means of a low-pass filter. A cut-off frequency F of the low-pass filter and a sound velocity V and a space D of the microphone pair have the following relationship.

$$F=V/2D$$

It has been known that arrival of a sound having a higher frequency than the cut-off frequency F causes a dip to be generated on an "8" shaped direction characteristic shown in FIG. 1A so that a directivity loses an "8" shape and an accurate direction of the sound source cannot be detected. For this reason, a frequency component which is higher than the cut-off frequency F is cut by means of the low-pass filter.

A processing of an output signal of the microphone pair will be described below with reference to FIG. 4.

In FIG. 4, **1**a and **1**b denote microphones forming the microphone pair **1** shown in FIG. 3A. Similarly, **2**a and **2**b to **7**a and **7**b denote microphones forming the microphone pairs **2** to **7**.

11 to **17** denote microphone pair detecting circuits having the same structure in which outputs of the microphones **1**a and **1**b to the microphones **7**a and **7**b are input and circuits **11**a to **11**f are provided.

11a denotes a subtraction circuit (SUB) acting as first calculating means, lle denotes an addition circuit (ADD) acting as second calculating means, **11**b denotes a low-pass filter (LPF), llc denotes a peak hold circuit (PKH), lid denotes an analog/digital converter (A/D), and **11**f denotes a switch (SW).

20 denotes a MPU (microprocessor unit) acting as a host unit for performing a signal processing. **21** denotes an addition circuit (ADD) for adding a plurality of input signals. **22** denotes a RS232C driver of a low-speed interface.

Furthermore, the circuits **11**c, **11**d and **20** correspond to means for detecting information about a direction of a sound source.

For example, respective outputs of sounds received by the microphones (M) **1**a and **1**b are input to the SUB **11**a for subtracting the outputs and the ADD **11**e for adding the outputs. The SUB **11**a and the ADD **11**e are formed by using an operational amplifier according to the prior art.

A subtraction output sent from the SUB **11**a (which corresponds to a difference signal) is input to the LPF **11**b having the cut-off frequency F (F=V/2D). An output of the LPF **11**b is held at a maximum value by means of the PKH **11**c.

The maximum value held by the PKH **11**c is converted into analog/digital conversion data (A/DDATA) by the A/D lid, and is input to the MPU **20**.

Similarly, outputs of the microphones **2a** and **2b** to the microphones **7a** and **7b** are input to the microphone pair detecting circuits **12** to **17** to obtain six A/DDATAs. The A/DDATAs are input to the MPU **20**, respectively.

In the MPU **20**, values of seven A/DDATAs are judged. Based on a minimum value, a direction signal is generated and is output as the detected direction signal of a sound source through the RS232C driver **22**.

Furthermore, the MPU **20** sends an output for MIC selection for selecting an addition output (a sum signal) of A/DDATA having a minimum value and an addition output (a sum signal) corresponding to second and third smallest values . . . of the A/DDATA if necessary on the basis of a result of the judgment of values of the seven A/DDATAs , thereby turning on the SW **11f**. These sum signals are caused to pass and are added to the ADD **21**.

If it is decided that the second and third smallest values of the seven A/DDATAs are equal to each other, the SW **11f** is controlled to select one of them.

In the ADD **21**, the SWs **11f** are turned on to improve the S/N ratio by using at least one of the sum signals which have passed. Then, the sum signal is sent, to a next stage, as a desired detection signal of the microphone having the improved S/N ratio.

While the number of the microphone pairs that are to be added is generally determined by a frequency of the microphone which is to be detected, an arrangement angle of the microphone pairs and a space between the microphone pairs, detailed description will be omitted.

The S/N ratio is improved by 3 dB by addition of one microphone output. Therefore, a detection signal of the microphone pair which gives a minimum value of a difference signal is improved by 3 dB at the maximum as compared with a detection signal of one microphone.

Accordingly, the S/N ratio can be improved by 6 dB at the maximum by adding the detection signals of the microphone pairs corresponding to the minimum and second smallest values of the difference signal, respectively.

[Second Embodiment]

Description will be given with reference to FIGS. **5A** and **5B**. FIGS. **5A** and **5B** show structures of microphone arrangement according to a second embodiment.

In a first example shown in FIG. **5A**, eight microphones are arranged in two lines.

The microphone pairs are combined in seven ways, that is, four ways of the microphones **1a** and **1b** to the microphones **1a** and **4b**, and three ways of the microphones **1b** and **2a** to the microphones **1b** and **4a**.

By selecting the seven ways of combination, it is possible to detect a direction of a sound source and to improve a S/N ratio of an object sound in the same manner as in the embodiment shown in FIG. **3A**. In this case, the number of the required microphones can be reduced to 8, which is smaller than the embodiment shown in FIG. **3A**.

The sound source is detected within a range from a direction orthogonal to a direction of arrangement of the microphones **1b** and **4a** to a direction orthogonal to a direction of arrangement of the microphones **1a** and **4b**.

In a second example shown in FIG. **5B**, four microphones **1a** to **4a** are arranged in a line, and two microphones **1b** and **1c** are arranged on both sides of the microphone **1a**.

The microphone pairs are combined in seven ways, that is, four ways of the microphones **1b** and **1a** to the microphones **1b** and **4a**, and three ways of the microphones **1c** and **2a** to the microphones **1c** and **4a**, for example.

In the same manner as in the first example, the direction of the sound source can be detected and the S/N ratio of the object sound can be improve by selecting the seven ways of combination as in the embodiment shown in FIG. **3A**. In this case, the number of the required microphones can be reduced to 6, which is smaller than the embodiment shown in FIG. **3A** and the first example.

The sound source is detected within a range from a direction orthogonal to a direction of arrangement of the microphones **1c** and **4a** to a direction orthogonal to a direction of arrangement of the microphones **1b** and **4a**.

[Third Embodiment]

FIG. **6** is a diagram showing a structure of a microphone pair according to a third embodiment. In the present embodiment, there will be shown an example of a microphone array in which two pairs of microphones orthogonal to each other form a set which is arranged at a crossing angle of 45 degrees or less.

According to the present embodiment, a direction of a sound source and a distance to the sound source are detected based on the microphone array in which 16 microphones are arranged on a concentric circle at an angle of 22.5 degrees as shown in FIG. **6**.

More specifically, microphones **N1a** and **N1b** form a microphone pair **N1**, microphones **N2a** and **N2b** form a microphone pair **N2**, microphones **N3a** and **N3b** form a microphone pair **N3**, microphones **N4a** and **N4b** form a microphone pair **N4**, microphones **N5a** and **N5b** form a microphone pair **N5**, microphones **N6a** and **N6b** form a microphone pair **N6**, microphones **N7a** and **N7b** form a microphone pair **N7**, and microphones **N8a** and **N8b** form a microphone pair **N8**.

Referring to the set of microphones, the microphone pair **N1** and the microphone pair **N5** orthogonal thereto form microphone sets **N1** and **N5**, the microphone pair **N2** and the microphone pair **N6** orthogonal thereto form microphone sets **N2** and **N6**, the microphone pair **N3** and the microphone pair **N7** orthogonal thereto form microphone sets **N3** and **N7**, and the microphone pair **N4** and the microphone pair **N8** orthogonal thereto form microphone sets **N4** and **N8**.

The microphone sets **N2** and **N6** are arranged on a concentric circle at an angle of 22.5 degrees with respect to the microphone sets **N1** and **N5**, the microphone sets **N3** and **N7** are arranged at an angle of 22.5 degrees with respect to the microphone sets **N2** and **N6**, and the microphone sets **N4** and **N8** are arranged at an angle of 22.5 degrees with respect to the microphone sets **N3** and **N7**. Accordingly, a crossing angle of the microphone pairs is 22.5 degrees in the present embodiment.

It is assumed that a position of a sound source S and that of the microphone array have a relationship shown in FIG. **6**. More specifically, it is assumed that the position of the sound source S is set on an extension line of the microphone pair **N1** on the same plane as the microphone array.

In the case where the sound source S is set in such a position, the direction of the sound source can be detected by the microphone pair **N5** having the smallest difference output.

A method for detecting the distance to the sound source will be described below.

A sound volume is inversely proportional to the distance from the sound source. Therefore, if a space between the microphone pairs is represented by 2R (that is, a radius of the microphone array is represented by R), and a distance

from a central position of the microphone array to the sound source S is represented by L, a ratio of sound pressure outputs detected by the microphone pair N1 orthogonal to the microphone pair N5 has the following relationship, wherein a sound pressure output of the microphone N1*a* is represented by N1*a*OUT and that of the microphone N1*b* is represented by N1*b*OUT.

$$N1aOUT/N1bOUT=(L+R)/(L-R)$$

the distance L from the central position of the microphone array to the sound source S can be obtained by the following equation.

$$L=(N1aOUT+N1bOUT)R/(N1aOUT-N1bOUT)$$

A non-directional microphone having a constant sensitivity to a direction of 360 degrees is used for the microphone array. By using a method shown in FIG. 7, the sensitivity of the microphone is adjusted.

The sound source S is provided on a central axis of the microphone array in such a manner that a distance from the sound source S to each microphone is constant. Thus, the sensitivity of each microphone is adjusted such that an output thereof is identical.

FIG. 8 is a block diagram showing a signal processing circuit. A signal processing will be described below with reference to the block diagram.

In FIG. 8, 31 to 38 denote microphone pair detecting circuits corresponding to the microphone pairs N1 to N8, respectively. Since each microphone pair detecting circuit is identical, only an internal portion of the microphone pair detecting circuit 31 is shown.

Data output from the microphone pair detecting circuits 31 to 38 are input to a MPU 20 and are output from the MPU 20 through a RS232C driver 22 in the same manner as in the circuit shown in FIG. 4. The MPU 20 is provided with a ROM 23 and a RAM 24.

Since the microphone pair detecting circuits 31 to 38 have the same function, the microphone pair detecting circuit 31 will be described as an example.

An output of the microphone pair N1, that is, an output of each of microphones N1*a* and N1*b* is input to an amplifier 11*g* indicated at AMP and is amplified, and is input to a subtraction circuit 11*a* indicated at SUB. The subtraction circuit 11*a* is formed by using an operational amplifier and the like according to the prior art.

An output of the subtraction circuit 11*a* is input to a low-pass filter 11*b* having a specific cut-off frequency F which is indicated at LPF.

As described above, the cut-off frequency F of the low-pass filter 11*b* and a sound velocity v and a space D (=2R) of the microphone pair have a relationship of F=V/2D. If a sound having a higher frequency than the cut-off frequency F arrives, an accurate direction of a sound source cannot be detected. Therefore, a frequency component which is higher than the cut-off frequency F is cut by means of the low-pass filter 11*b*.

An output of the low-pass filter 11*b* is held at a maximum value by a peak hold circuit 11*c* indicated at PKH. The held maximum value is converted into digital subtraction data by an analog/digital converter 11*d* indicated at A/D and is input to the MPU 20.

Each output of the amplifier 11*g* is held at a maximum value by a peak hold circuit 11*h* indicated at PKH. The held maximum value is converted into digital data by an analog/digital converter 11*i* indicated at A/D, and is input to the MPU 20.

The MPU 20 compares all the subtraction data of the microphone pairs N1 to N8, and selects the microphone pair having the minimum subtraction data to be stored in the RAM 24. Consequently, a direction of the sound source S is detected.

Next, any of the microphone pairs N1 to N8 which is orthogonal to the microphone pair having the minimum subtraction data is selected. Output data of the selected microphone pair is stored in the RAM.24. Based on the output data of the microphone pair, a distance L from a central position of the microphone array to the sound source S is calculated by the above-mentioned equation.

Such an operation program is stored in the ROM 23. The distance L is calculated in accordance with the stored program.

The MPU 20 sends data on the detected direction of the sound source S and data on the distance to a back processor such as a personal computer through the RS232C driver 22.

Thus, the direction of the sound source and the distance to the sound source can be detected by using a microphone array in which cross-shaped microphone sets having two microphone pairs orthogonal to each other are arranged on a concentric circle at an angle of 22.5 degrees.

While four sets of microphones are arranged on the concentric circle and a crossing angle of the microphone pairs is 22.5 degrees in the present embodiment, the microphone sets may be arranged at a crossing angle of 45 degrees or less. More specifically, if the condition that the crossing angle is 45 or less is met, a plurality of microphone sets may be arranged on the concentric circle at regular intervals. For example, two sets of microphones may be arranged at a crossing angle of 45 degrees, three sets of microphones may be arranged at a crossing angle of 30 degrees, or five sets of microphones may be arranged at a crossing angle of 18 degrees.

While only one microphone array has been used in the above description, two microphone arrays capable of detecting the direction of the sound source S can used to calculate the direction of the sound source S and the distance to the sound source S, which will be described below in a fourth embodiment.

[Fourth Embodiment]

FIG. 9 is a diagram showing a structure of a microphone pair according to a fourth embodiment. In the present embodiment, two microphone arrays are used as an example of arrangement.

In the present embodiment, two microphone arrays 41 and 42 are arranged apart from each other by a distance r. It is possible to use the microphone array shown in FIG. 3A and the microphone array shown in the [first example] or [second example] of FIGS. 5A and 5B. The microphone array shown in FIG. 6 may be used.

As shown in FIG. 9, a distance between two microphone arrays 41 and 42 is represented by r, a distance from microphone array faces formed by the microphone arrays 41 and 42 to a sound source S is represented by L, a distance from the microphone array 42 to a central position of the microphone array face is represented by H, and a direction of the sound source S detected by the microphone array 41 and that of the sound source S detected by the microphone array 42 are represented by $\theta 1$ and $\theta 2$. Consequently, the following equations are obtained.

$$L/(r+H)=\tan\theta 1$$

$$L/H=\tan\theta 2$$

Consequently, the distance L from the microphone array face to the sound source S can be calculated by the following equation.

$$L=r(\tan\theta 1/(1-\tan\theta 1/\tan\theta 2))$$

The distance H from the microphone array **42** to the central position of the microphone array face can be calculated by the following equation.

$$H=r(\tan\theta 1/(\tan\theta 2-\tan\theta 1))$$

The direction of the sound source S can be calculated by the following equation.

$$\tan\theta=1/(r/2L+1/\tan\theta 2)$$

The distance r between the microphone arrays **41** and **42** which has previously been stored is used. The distance L or H which is longer is selected as the distance from the microphone array to the sound source S.

FIG. **10** is a diagram showing a circuit structure in which two microphone arrays are used. In FIG. **10**, **43** and **44** denote sound source direction discriminators, each including a microphone array, a microphone pair detecting circuit and a MPU, and **45** denotes an arithmetic unit.

The circuit shown in FIG. **4** can be used as the sound source direction discriminators **43** and **44**. The sound source direction discriminators **43** and **44** detect direction data θ1 and θ2 of the sound source S from an output of the microphone array, and output the same data to the arithmetic unit **45**.

In the arithmetic unit **45**, the distance L from the microphone array face to the sound source S or the distance H from the microphone array to the central position of the microphone array face and the direction θ of the sound source are calculated based on the direction data θ1 and θ2 of the sound source S and the distance r between two microphone arrays which have previously been stored. The data is output to a back processor (host unit) such as a personal computer.

Thus, the direction of the sound source and the distance to the sound source can be detected by using two microphone arrays.

A technique for detecting the direction of the sound source and the distance to the sound source can be utilized for software for producing a communication between a personal computer and a sounder (talker) positioned before the personal computer, for example. More specifically, the technique can be utilized for various kinds of communication software in which men or animals such as birds are displayed on a screen and they are turned in a direction of a sound source generated by the sounder.

As is apparent from the above description, the present invention produces the effect that a direction of a sound source can be detected with a small and simple structure and an object sound can be extracted with a high S/N ratio.

Although the present invention has fully been described by way of example with reference to the accompanying drawings, it is to be understood that various changes and modifications will be apparent to those skilled in the art. Therefore, unless otherwise such changes and modifications depart from the scope of the invention, should be construed as being included therein.

What is claimed is:

1. A microphone system comprising:

a plurality of microphone pairs, each pair having two microphones arranged apart from each other at a predetermined space at a crossing angle of 60 degrees or less;

a plurality of first calculating means for calculating a difference signal of outputs of each microphone pair;

a plurality of second calculating means for calculating a sum signal of outputs of each microphone pair;

means for detecting, as sound source direction information, a minimum value output from each output of the first calculating means; and

means for selecting a sum signal of the microphone pair corresponding to the minimum value output and outputting the selected sum signal as sound information.

2. The microphone system according to claim **1**, wherein a sum signal corresponding to a second smallest difference signal is also selected and output in addition to the sum signal corresponding to the minimum value of the difference signal.

3. The microphone system according to claim **1**, wherein a plurality of microphone pairs are arranged on a concentric circle.

4. The microphone system according to claim **1**, further comprising a low-pass filter for filtering the difference signal output from the first calculating means at a cut-off frequency F represented by V/2D=F between a sound velocity V and the space D.

5. A microphone system comprising:

a microphone array in which microphone sets are arranged at a crossing angle of 45 degrees or less, each microphone set including a first microphone pair having two microphones arranged apart from each other at a predetermined space and a second microphone pair orthogonal to the first microphone pair;

a plurality of first calculating means for calculating a difference signal of outputs of each microphone pair;

means for detecting, as a direction of a sound source, a minimum value output from each output of the first calculating means;

second calculating means for calculating a ratio of output voltages of a microphone pair orthogonal to a microphone pair corresponding to the minimum value output; and

means for calculating a distance from the microphone array to a sound source based on the ratio of the output voltages which is calculated by the second calculating means.

6. A microphone system comprising a plurality of sound source direction discriminators including a plurality of microphone pairs, each pair having two microphones

arranged apart from each other at a predetermined space at a crossing angle of 60 degrees or less, a plurality of calculating means for calculating a difference signal of outputs of each microphone pair, and means for detecting, as a direction of a sound source, a minimum value output from each output of the calculating means, wherein a crossing point of the directions of the sound source which are obtained by the sound source direction discriminators is calculated, thereby detecting a position of the sound source.

7. The microphone system according to any of claims **1** to **6**, wherein the microphone is a non-directional microphone.

8. The microphone system according to claim **7**, wherein the sound source is provided on a central axis of a microphone array, thereby adjusting a sensitivity of each microphone pair.

* * * * *

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(12) **United States Patent**
Elko et al.

(10) **Patent No.:** **US 6,584,203 B2**
(45) **Date of Patent:** **Jun. 24, 2003**

(54) **SECOND-ORDER ADAPTIVE DIFFERENTIAL MICROPHONE ARRAY**

(75) Inventors: **Gary W. Elko**, Summit, NJ (US); **Heinz Teutsch**, Nuremberg (DE)

(73) Assignee: **Agere Systems Inc.**, Allentown, PA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/999,298**

(22) Filed: **Oct. 30, 2001**

(65) **Prior Publication Data**

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Related U.S. Application Data

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(51) **Int. Cl.**[7] **H04R 3/00**; H04R 5/00; H04B 15/00

(52) **U.S. Cl.** **381/92**; 381/94.6; 381/26

(58) **Field of Search** 381/92, 94.6, 26

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Primary Examiner—Forester W. Isen
Assistant Examiner—Elizabeth McChesney
(74) *Attorney, Agent, or Firm*—Steve Mendelsohn

(57) **ABSTRACT**

A second-order adaptive differential microphone array (ADMA) has two first-order elements (e.g., **802** and **804** of FIG. 8), each configured to convert a received audio signal into an electrical signal. The ADMA also has (i) two delay nodes (e.g., **806** and **808**) configured to delay the electrical signals from the first-order elements and (ii) two subtraction nodes (e.g., **810** and **812**) configured to generate forward-facing and backward-facing cardioid signals based on differences between the electrical signals and the delayed electrical signals. The ADMA also has (i) an amplifier (e.g., **814**) configured to amplify the backward-facing cardioid signal by a gain parameter; (ii) a third subtraction node (e.g., **816**) configured to generate a difference signal based on a difference between the forward-facing cardioid signal and the amplified backward-facing cardioid signal; and (iii) a lowpass filter (e.g., **818**) configured to filter the difference signal from the third subtraction node to generate the output signal for the second-order ADMA. The gain parameter for the amplifier can be adaptively adjusted to move a null in the back half plane of the ADMA to track a moving noise source. In a subband implementation, a different gain parameter can be adaptively adjusted to move a different null in the back half plane to track a different moving noise source for each different frequency subband.

22 Claims, 12 Drawing Sheets

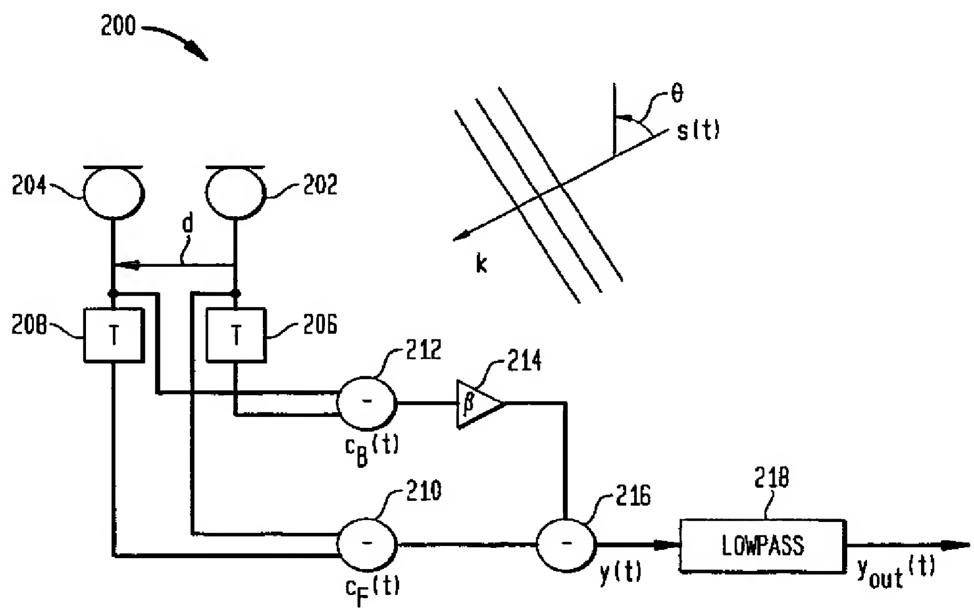

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FIG. 1

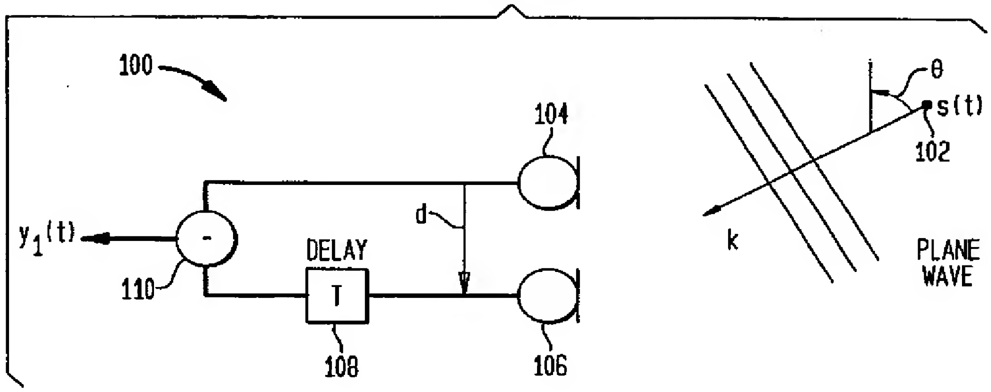


FIG. 2

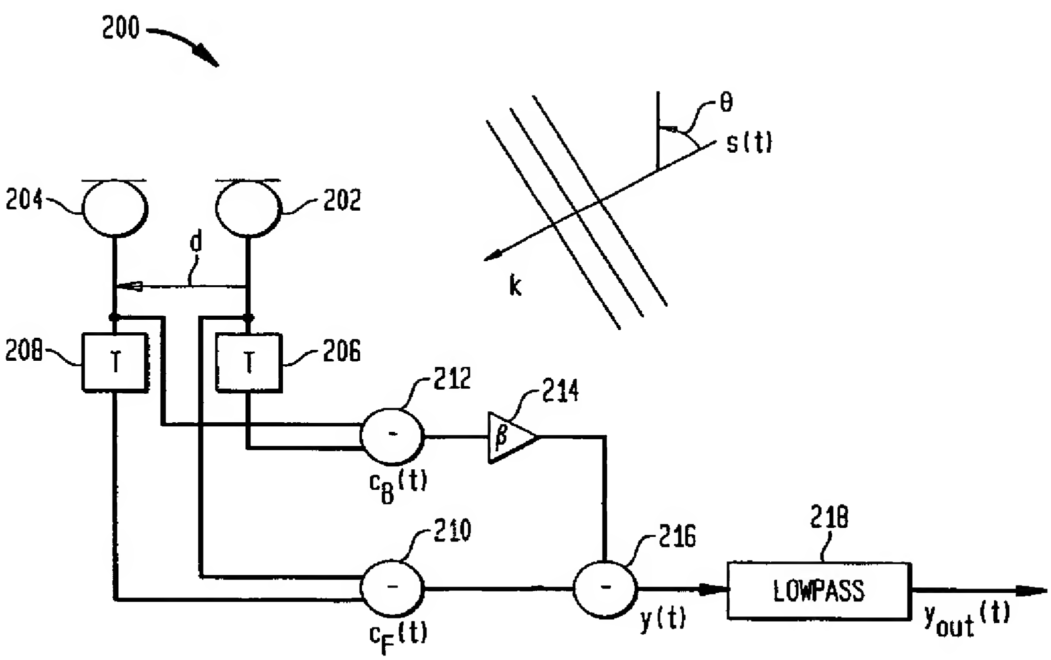

200
204
202
d
208
T
T
206
212
214
β
$c_B(t)$
210
216
218
LOWPASS
$c_F(t)$
$y(t)$
$y_{out}(t)$
θ
s(t)
k

FIG. 3

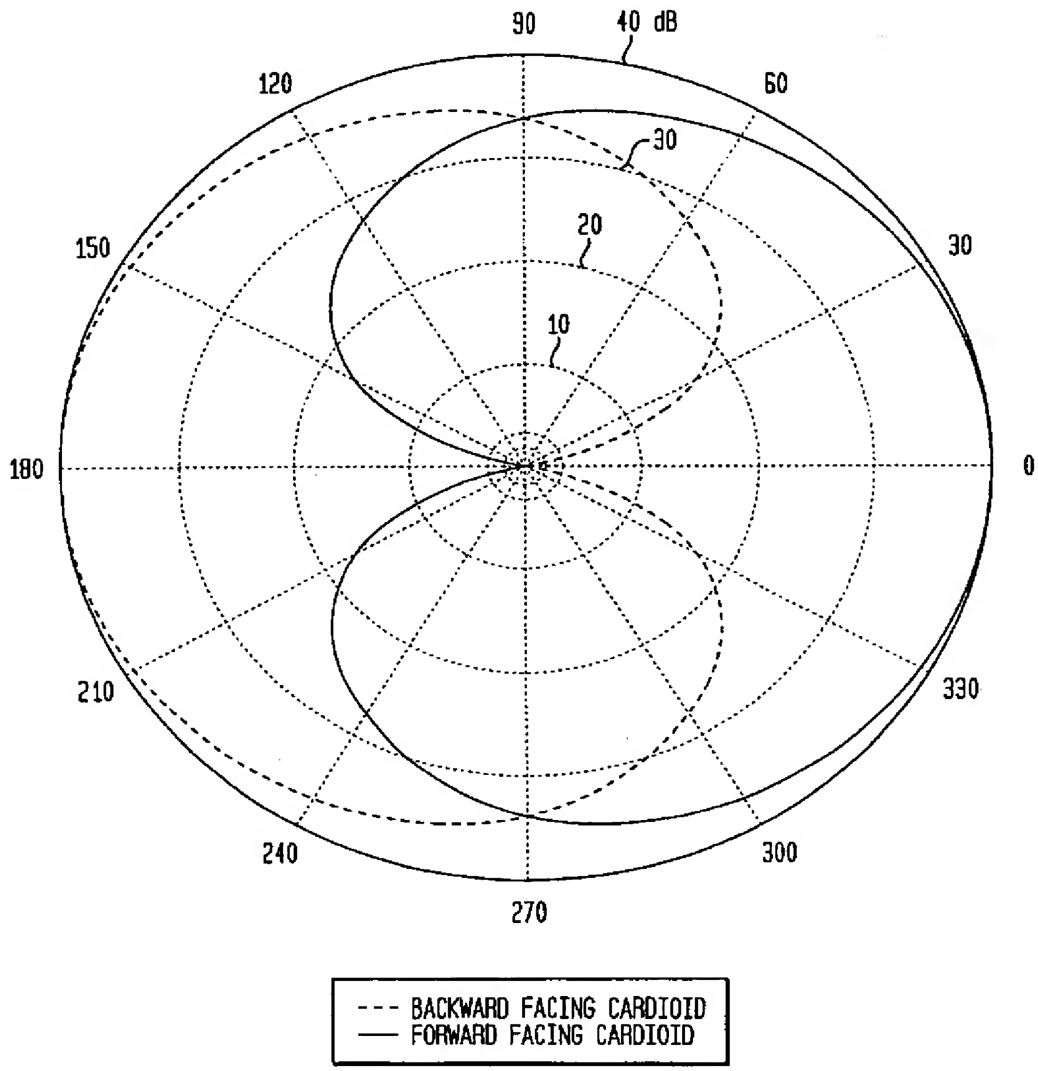
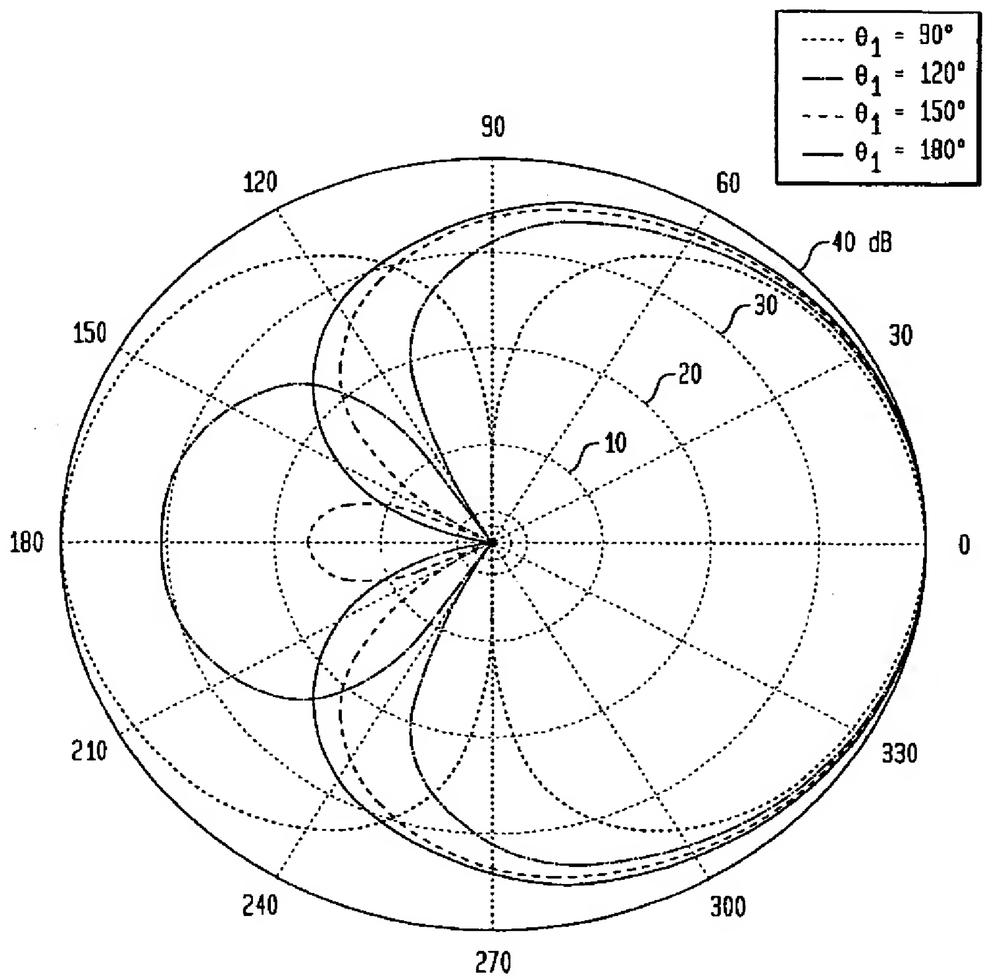

90
40 dB
60
120
30
20
10
150
30
180
0
210
330
240
300
270
BACKWARD FACING CARDIOID
FORWARD FACING CARDIOID

FIG. 4



---- θ1 = 90°
—— θ1 = 120°
- - - θ1 = 150°
—— θ1 = 180°
90
60
30
0
330
300
270
240
210
180
150
120
40 dB
30
20
10

FIG. 5

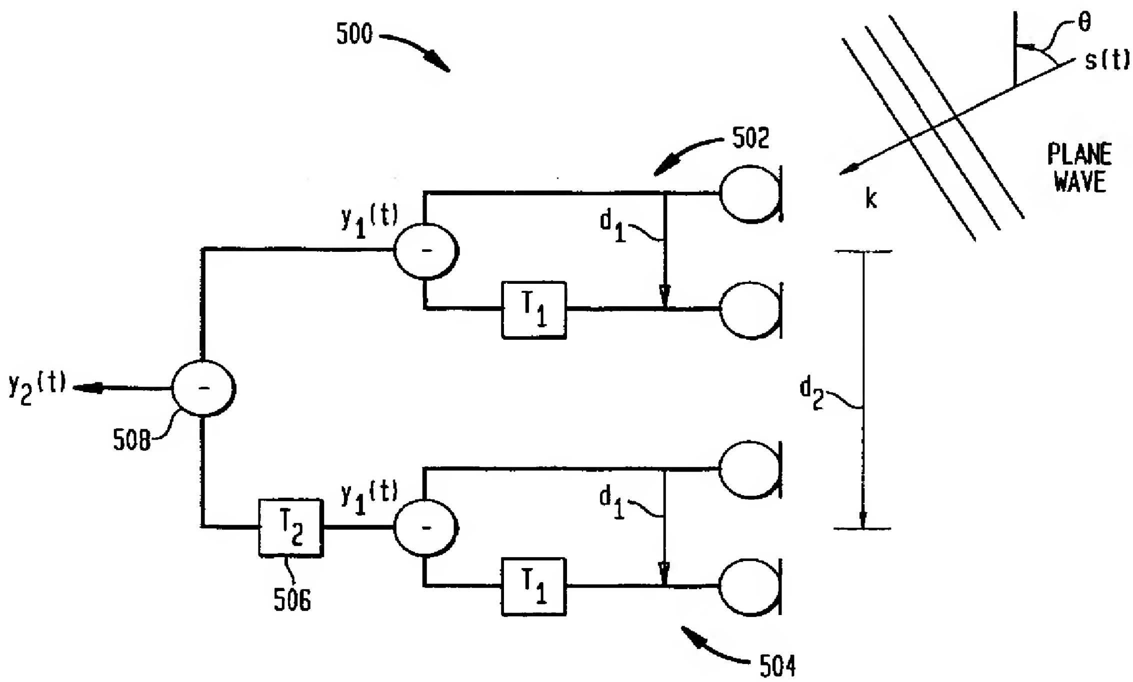


FIG. 6

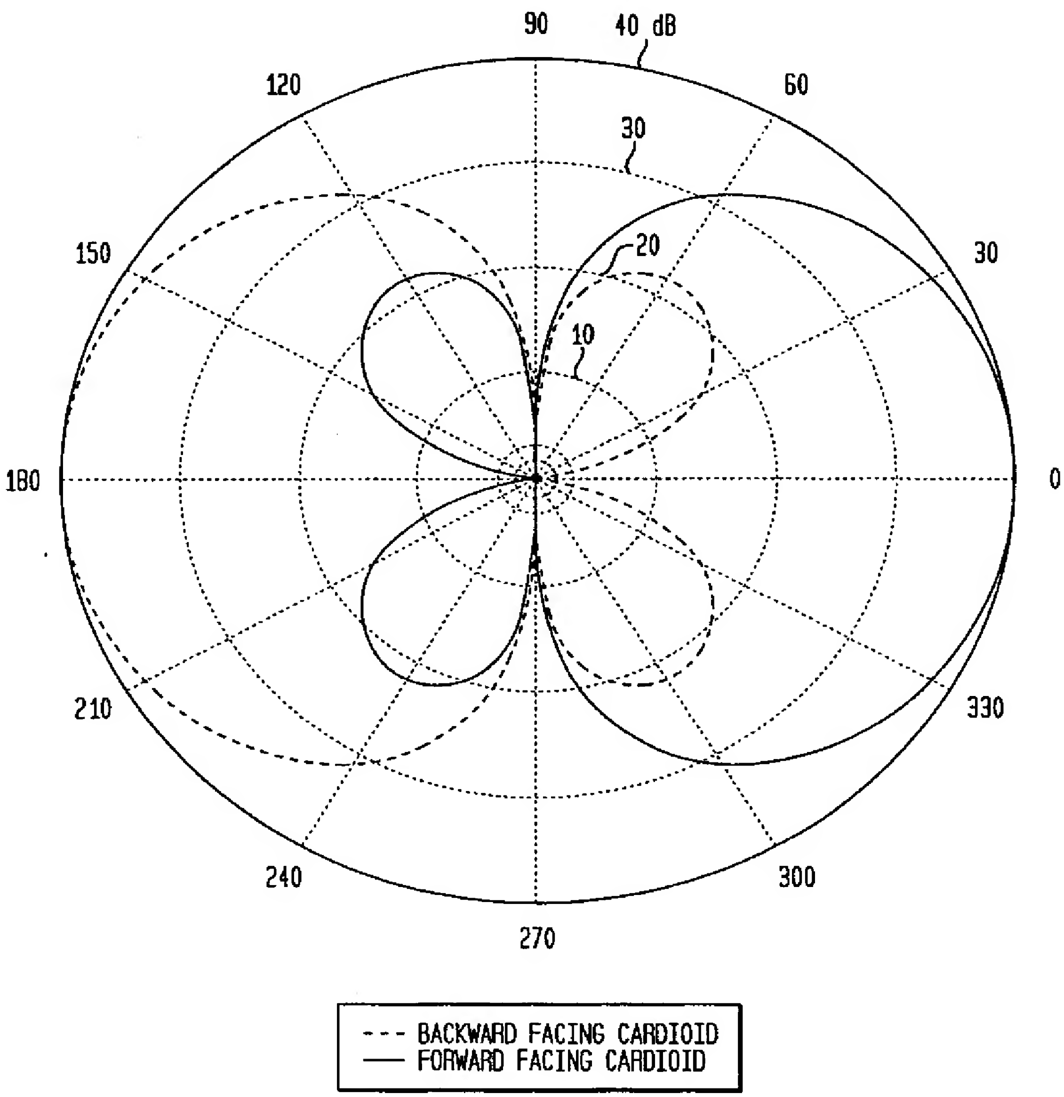
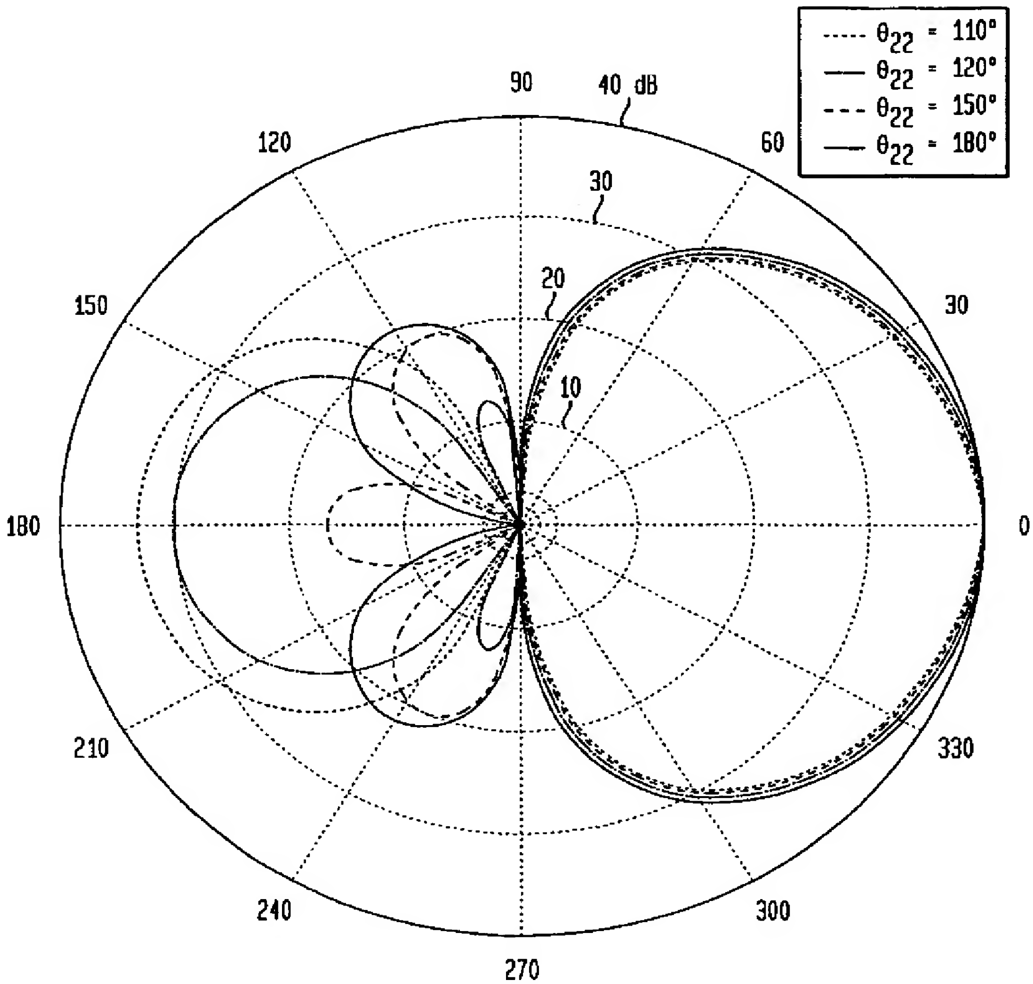

90
40 dB
120
60
30
150
20
30
10
180
0
210
330
240
300
270
BACKWARD FACING CARDIOID
FORWARD FACING CARDIOID

FIG. 7



θ22 = 110°
θ22 = 120°
θ22 = 150°
θ22 = 180°
90
40 dB
120
60
30
20
10
150
30
180
0
210
330
240
300
270

FIG. 8

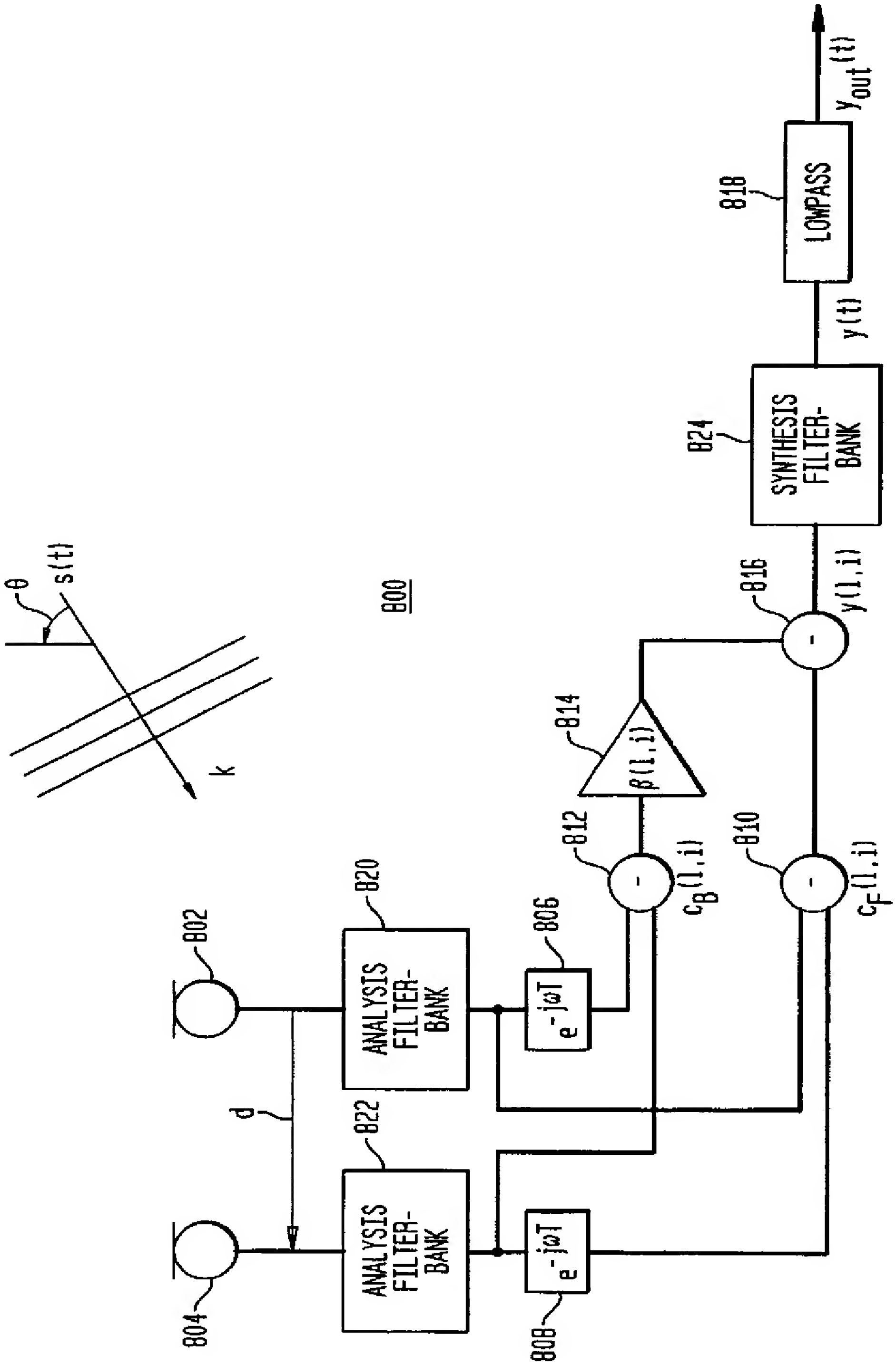
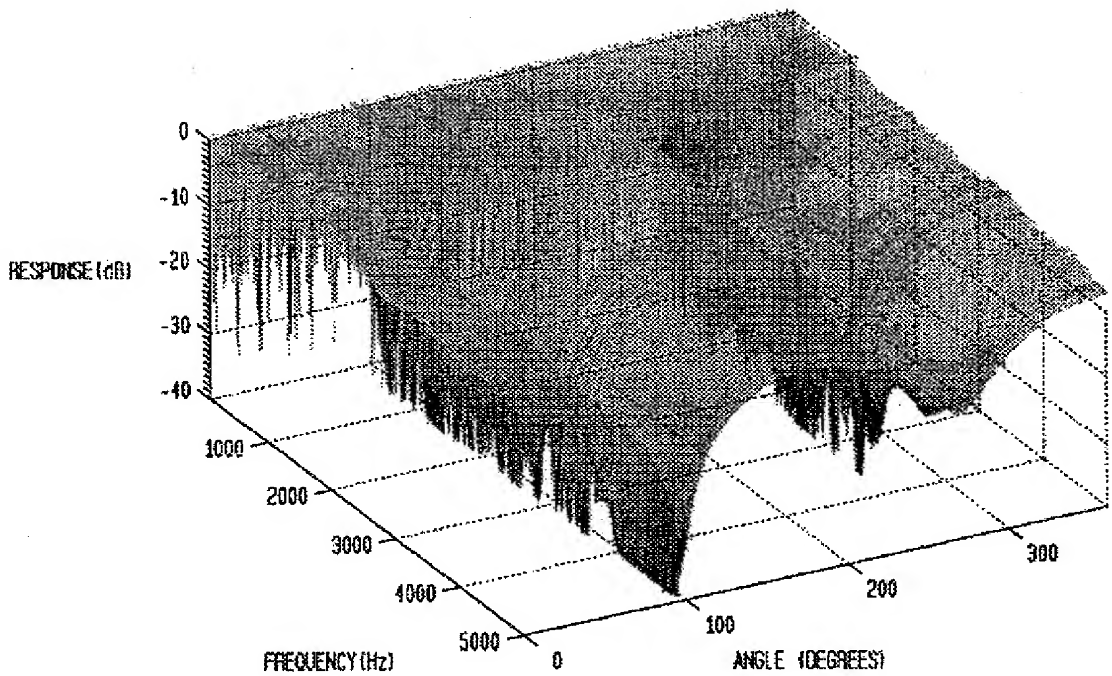
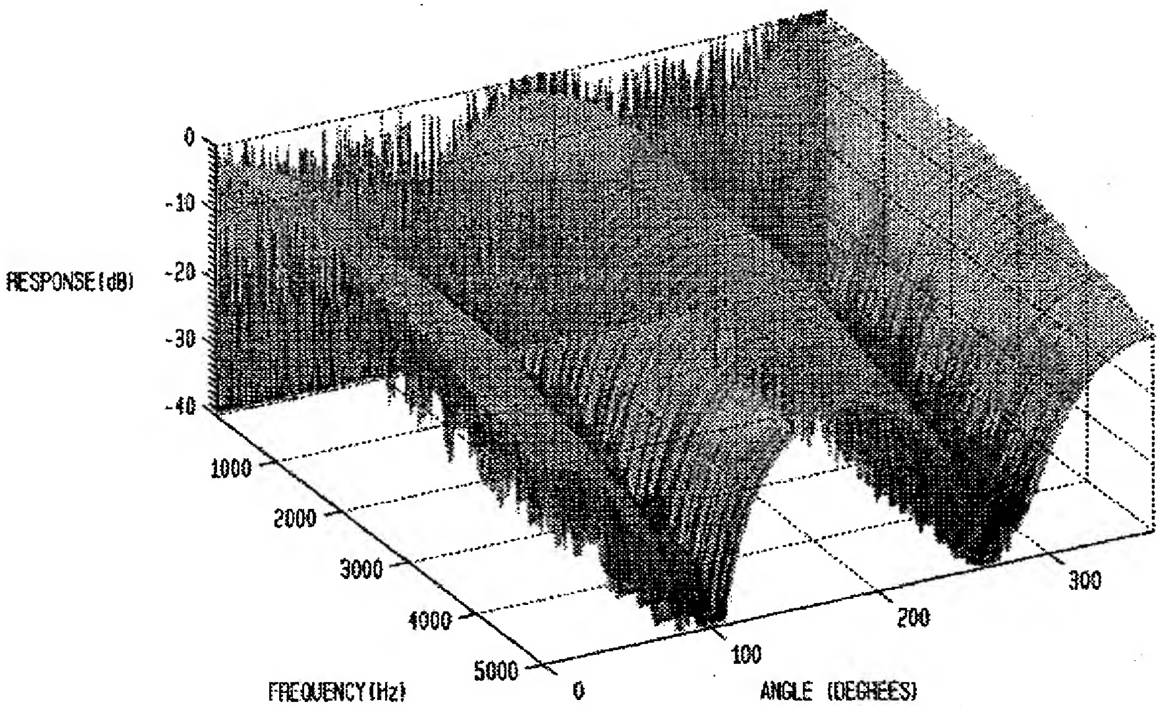

θ
s(t)
k
800
802
804
d
820
822
ANALYSIS FILTER-BANK
ANALYSIS FILTER-BANK
806
808
$e^{-j\omega T}$
$e^{-j\omega T}$
812
810
$c_B(l,i)$
$c_F(l,i)$
814
$\beta(l,i)$
816
$y(l,i)$
824
SYNTHESIS FILTER-BANK
$y(t)$
818
LOWPASS
$y_{out}(t)$

FIG. 9A



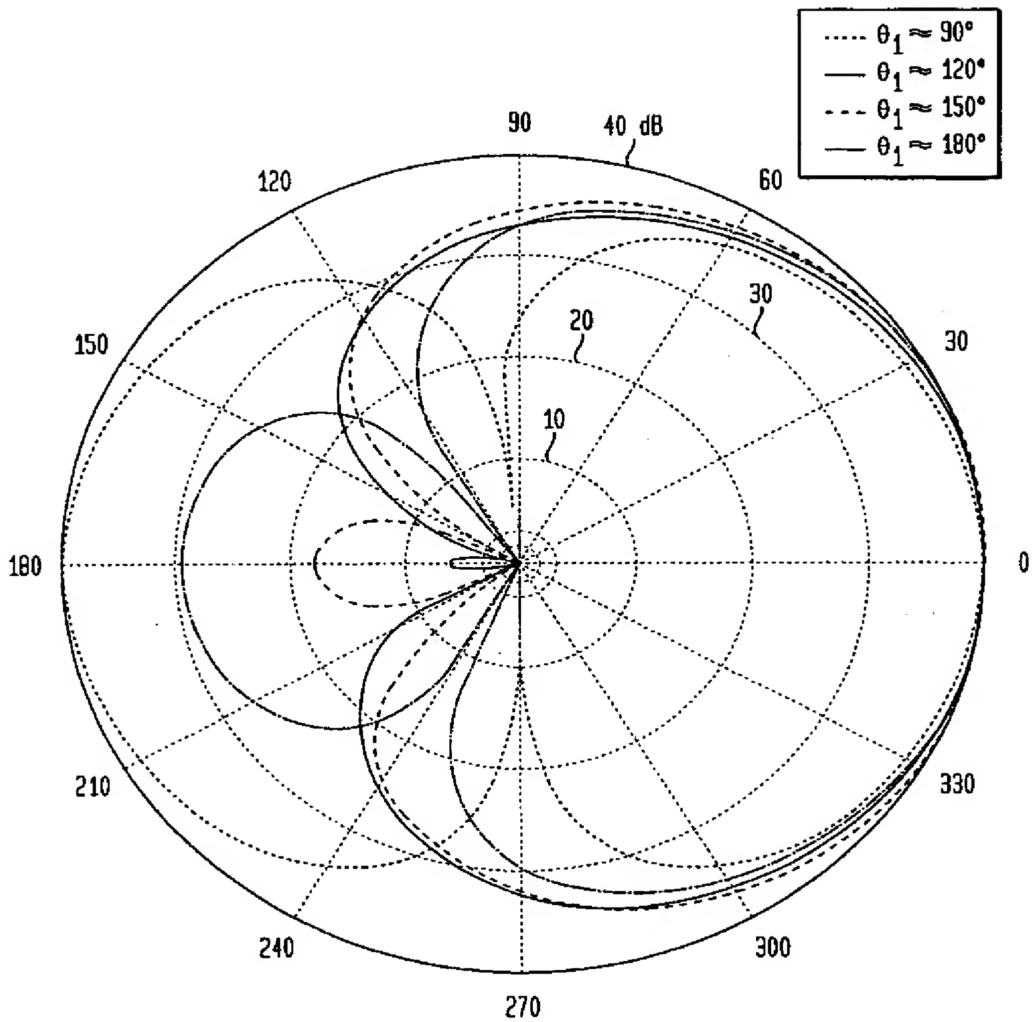
0
-10
-20
-30
-40
RESPONSE (dB)
1000
2000
3000
4000
5000
FREQUENCY (Hz)
0
100
200
300
ANGLE (DEGREES)

FIG. 9B



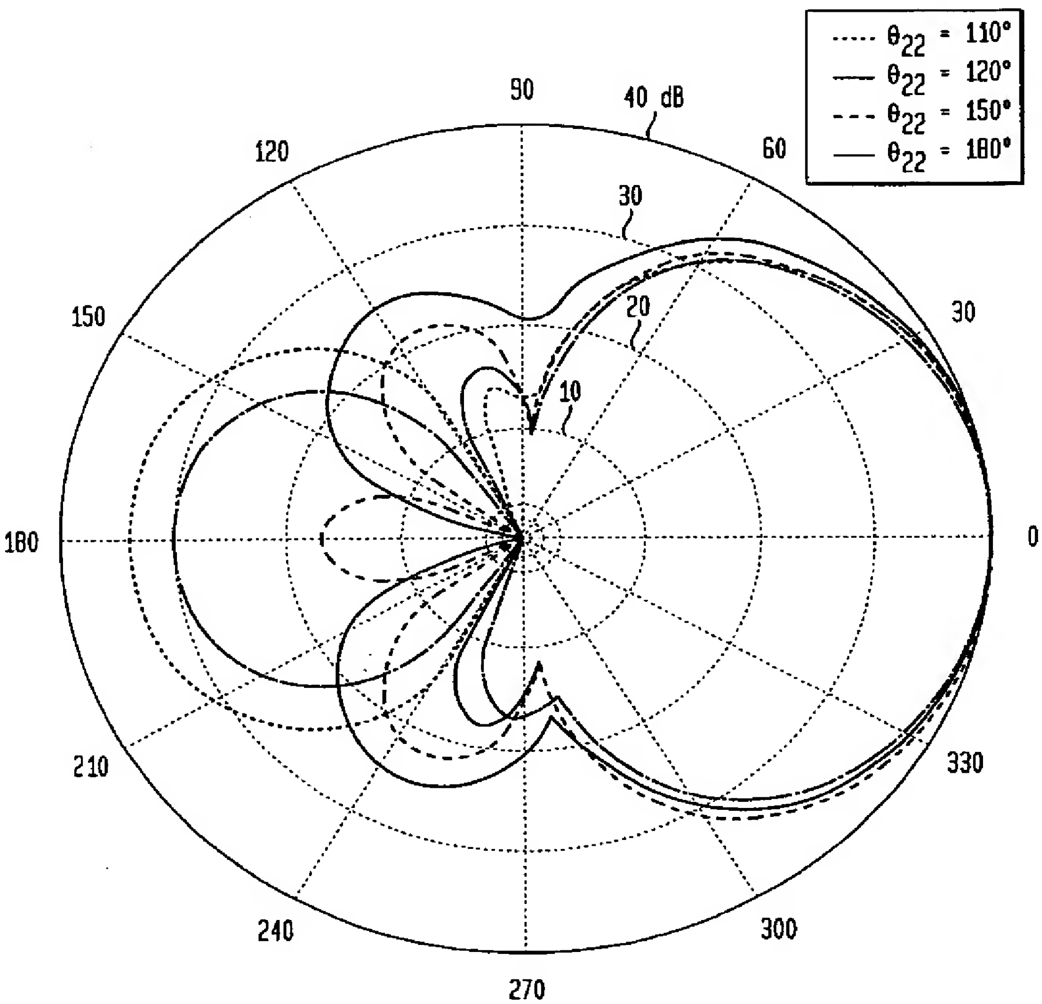
RESPONSE(dB)
0
-10
-20
-30
-40
1000
2000
3000
4000
5000
FREQUENCY(Hz)
0
100
200
300
ANGLE (DEGREES)

FIG. 10



θ1 ≈ 90°
θ1 ≈ 120°
θ1 ≈ 150°
θ1 ≈ 180°
90
40 dB
120
60
150
30
20
30
10
180
0
210
330
240
300
270

FIG. 11



θ22 = 110°
θ22 = 120°
θ22 = 150°
θ22 = 180°
40 dB
30
20
10
90
60
30
0
330
300
270
240
210
180
150
120

SECOND-ORDER ADAPTIVE DIFFERENTIAL MICROPHONE ARRAY

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of the filing date of U.S. provisional application No. 60/306,271, filed on Jul. 18, 2001.

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to microphone arrays that employ directionality characteristics to differentiate between sources of noise and desired sound sources.

2. Description of the Related Art

The presence of background noise accompanying all kinds of acoustic signal transmission is a ubiquitous problem. Speech signals especially suffer from incident background noise, which can make conversations in adverse acoustic environments virtually impossible without applying appropriately designed electroacoustic transducers and sophisticated signal processing. The utilization of conventional directional microphones with fixed directivity is a limited solution to this problem, because the undesired noise is often not fixed to a certain angle.

SUMMARY OF THE INVENTION

Embodiments of the present invention are directed to adaptive differential microphone arrays (ADMAs) that are able to adaptively track and attenuate possibly moving noise sources that are located in the back half plane of the array. This noise attenuation is achieved by adaptively placing a null into the noise source's direction of arrival. Such embodiments take advantage of the adaptive noise cancellation capabilities of differential microphone arrays in combination with digital signal processing. Whenever undesired noise sources are spatially non-stationary, conventional directional microphone technology has its limits in terms of interference suppression. Adaptive differential microphone arrays (ADMAs) with their null-steering capabilities promise better performance.

In one embodiment, the present invention is a second-order adaptive differential microphone array (ADMA), comprising (a) a first first-order element (e.g., **802** of FIG. **8**) configured to convert a received audio signal into a first electrical signal; (b) a second first-order element (e.g., **804** of FIG. **8**) configured to convert the received audio signal into a second electrical signal; (c) a first delay node (e.g., **806** of FIG. **8**) configured to delay the first electrical signal from the first first-order element to generate a delayed first electrical signal; (d) a second delay node (e.g., **808** of FIG. **8**) configured to delay the second electrical signal from the second first-order element to generate a delayed second electrical signal; (e) a first subtraction node (e.g., **810** of FIG. **8**) configured to generate a forward-facing cardioid signal based on a difference between the first electrical signal and the delayed second electrical signal; (f) a second subtraction node (e.g., **812** of FIG. **8**) configured to generate a backward-facing cardioid signal based on a difference between the second electrical signal and the delayed first electrical signal; (g) an amplifier (e.g., **814** of FIG. **8**) configured to amplify the backward-facing cardioid signal by a gain parameter to generate an amplified backward-facing cardioid signal; and (h) a third subtraction node (e.g., **816** of FIG. **8**) configured to generate a difference signal based on a difference between the forward-facing cardioid signal and the amplified backward-facing cardioid signal.

In another embodiment, the present invention is an apparatus for processing signals generated by a microphone array (ADMA) having (i) a first first-order element (e.g., **802** of FIG. **8**) configured to convert a received audio signal into a first electrical signal and (ii) a second first-order element (e.g., **804** of FIG. **8**) configured to convert the received audio signal into a second electrical signal, the apparatus comprising (a) a first delay node (e.g., **806** of FIG. **8**) configured to delay the first electrical signal from the first first-order element to generate a delayed first electrical signal; (b) a second delay node (e.g., **808** of FIG. **8**) configured to delay the second electrical signal from the second first-order element to generate a delayed second electrical signal; (c) a first subtraction node (e.g., **810** of FIG. **8**) configured to generate a forward-facing cardioid signal based on a difference between the first electrical signal and the delayed second electrical signal; (d) a second subtraction node (e.g., **812** of FIG. **8**) configured to generate a backward-facing cardioid signal based on a difference between the second electrical signal and the delayed first electrical signal; (e) an amplifier (e.g., **814** of FIG. **8**) configured to amplify the backward-facing cardioid signal by a gain parameter to generate an amplified backward-facing cardioid signal; and (g) a third subtraction node (e.g., **816** of FIG. **8**) configured to generate a difference signal based on a difference between the forward-facing cardioid signal and the amplified backward-facing cardioid signal.

BRIEF DESCRIPTION OF THE DRAWINGS

Other aspects, features, and advantages of the present invention will become more fully apparent from the following detailed description, the appended claims, and the accompanying drawings in which:

FIG. **1** shows a schematic representation of a first-order adaptive differential microphone array (ADMA) receiving an audio signal from a signal source at a distance where farfield conditions are applicable;

FIG. **2** shows a schematic diagram of a first-order fullband ADMA based on an adaptive back-to-back cardioid system;

FIG. **3** shows the directivity pattern of the first-order ADMA of FIG. **2**;

FIG. **4** shows directivity patterns that can be obtained by the first-order ADMA for θ_1, values of 90°, 120°, 150°, and 180°;

FIG. **5** shows a schematic diagram of a second-order fullband ADMA;

FIG. **6** shows the directivity pattern of a second-order back-to-back cardioid system;

FIG. **7** shows the directivity patterns that can be obtained by a second-order ADMA formed from two dipole elements for θ_{22} values of 90°, 120°, 150°, and 180°;

FIG. **8** shows a schematic diagram of a subband two-element ADMA;

FIGS. **9A** and **9B** depict the fullband ADMA directivity patterns for first-order and second-order arrays, respectively; and

FIGS. **10** and **11** show measured directivity of first- and second-order subband implementations of the ADMA of FIG. **8**, respectively, for four simultaneously playing sinusoids.

DETAILED DESCRIPTION

First-Order Fullband ADMA

FIG. **1** shows a schematic representation of a first-order adaptive differential microphone array (ADMA) **100** receiving audio signal s(t) from audio source **102** at a distance where farfield conditions are applicable. When farfield conditions apply, the audio signal arriving at ADMA **100** can be treated as a plane wave. ADMA **100** comprises two zeroth-order microphones **104** and **106** separated by a distance d . Electrical signals generated by microphone **106** are delayed by inter-element delay T at delay node **108** before being subtracted from the electrical signals generated by microphone **104** at subtraction node **110** to generate the ADMA output y(t). The magnitude of the frequency and angular dependent response $H_1(f, \theta)$ of first-order ADMA **100** for a signal point source at a distance where farfield conditions are applicable can be written according to Equation (1) as follows:

$$|H_1(f,\theta)| = \left|\frac{Y_1(f,\theta)}{S(f)}\right| = 1 - e^{-j(2\pi fT + k\cdot d)} = 2\sin\frac{2\pi f[T + (d\cos\theta)/c]}{2} \tag{1}$$

where $Y_1(f, \theta)$ is the spectrum of the ADMA output signal y(t), $S(f)$ is the spectrum of the signal source, k is the sound vector, $|k|=k=2\pi f/c$ is the wavenumber, c is the speed of sound, and d is the displacement vector between microphones **104** and **106**. As indicated by the term $Y_1(f, \theta)$, the ADMA output signal is dependent on the angle θ between the displacement vector d and the sound vector k as well as on the frequency f of the audio signal s(t).

For small element spacing and short inter-element delay ($kd \ll \pi$ and $T \ll \frac{1}{2}f$, Equation (1) can be approximated according to Equation (2) as follows:

$$|H_1(f,\theta)| \approx 2\pi f[T + (d\cos\theta)/c]. \tag{2}$$

As can be seen, the right side of Equation (2) consists of a monopole term and a dipole term (cosθ). Note that the amplitude response of the first-order differential array rises linearly with frequency. This frequency dependence can be corrected for by applying a first-order lowpass filter at the array output. The directivity response can then be expressed by Equation (3) as follows:

$$D_1(\theta) = \frac{T}{T + d/c} + \left(1 - \frac{T}{T + d/c}\right)\cos\theta. \tag{3}$$

Since the location of the source **102** is not typically known, an implementation of a first-order ADMA based on Equation (3) would need to involve the ability to generate any time delay T between the two microphones. As such, this approach is not suitable for a real-time system. One way to avoid having to generate the delay T directly in order to obtain the desired directivity response is to utilize an adaptive back-to-back cardioid system

FIG. **2** shows a schematic diagram of a first-order fullband ADMA **200** based on an adaptive back-to-back cardioid system. In ADMA **200**, signals from both microphones **202** and **204** are delayed by a time delay T at delay nodes **206** and **208**, respectively. The delayed signal from microphone **204** is subtracted from the undelayed signal from microphone **202** at forward subtraction node **210** to form the forward-facing cardioid signal $C_F(t)$. Similarly, the delayed signal from microphone **202** is subtracted from the undelayed signal from microphone **204** at backward subtraction node **212** to form the backward-facing cardioid signal $c_B(t)$, which is amplified by gain β at amplifier **214**. The signal y(t) is generated at subtraction node **216** based on the difference between the forward and amplified backward signals. The signal y(t) is then lowpass filtered at filter **218** to generate the ADMA output signal $y_{out}(t)$.

FIG. **3** shows the directivity pattern of the first-order back-to-back cardioid system of ADMA **200**. ADMA **200** can be used to adaptively adjust the response of the backward facing cardioid in order to track a possibly moving noise source located in the back half plane. By choosing T=d/c, the back-to-back cardioid can be formed directly by appropriately subtracting the delayed microphone signals.

The transfer function $H_1(f, \theta)$ of first-order ADMA **200** can be written according to Equation (4) as follows:

$$H_1(f,\theta) = \frac{Y_{out}(f,\theta)}{S(f)} = 2je^{-j\pi fT}\left(\sin\frac{kd(1+\cos\theta)}{2} - \beta\sin\frac{kd(1-\cos\theta)}{2}\right). \tag{4}$$

where $Y_{out}(f, \theta)$ is the spectrum of the ADMA output signal $y_{out}(t)$.

The single independent null angle θ_1 of first-order ADMA **200**, which, for the present discussion, is assumed to be placed into the back half plane of the array ($90° \leq \theta_1 \leq 180°$), can be found by setting Equation (4) to zero and solving for $\theta=\theta_1$, which yields Equation (5) as follows:

$$\theta_1 = \arccos\left(\frac{2}{kd}\arctan\left(\frac{\beta-1}{\beta+1}\tan\frac{kd}{2}\right)\right), \tag{5}$$

which for small spacing and short delay can be approximated according to Equation (6) as follows:

$$\theta_1 \approx \arccos\frac{\beta-1}{\beta+1}, \tag{6}$$

where $0 \leq \beta \leq 1$ under the constraint ($90° \theta_1 \leq 180°$). FIG. **4** shows the directivity patterns that can be obtained by first-order ADMA **200** for θ_1 values of 90°, 120°, 150°, and 180°.

In a time-varying environment, an adaptive algorithm is preferably used in order to update the gain parameter β. In one implementation, a normalized least-mean-square (NLMS) adaptive algorithm may be utilized, which is computationally inexpensive, easy to implement, and offers reasonably fast tracking capabilities. One possible real-valued time-domain one-tap NLMS algorithm can be written according to Equation 2 (7a) and (7b) as follows:

$$y(i) = c_F(i) - \beta(i)c_B(i), \tag{7a}$$

$$\beta(i+1) = \beta(i) + \frac{\mu}{\alpha + \|c_B(i)\|}c_B(i)y(i), \tag{7b}$$

where $c_F(i)$ and $c_B(i)$ are the values for the forward- and backward-facing cardioid signals at time instance i, μ is an adaptation constant where $0<\mu<2$, and α is a small constant where $\alpha>0$.

Further information on first-order adaptive differential microphone arrays is provided in U.S. Pat. No. 5,473,701 (Cezanne et al.), the teachings of which are incorporated herein by reference.

Second-Order Fullband ADMA

FIG. 5 shows a schematic diagram of a second-order fullband ADMA **500** comprising two first-order ADMAs **502** and **504**, each of which is an instance of first-order ADMA **100** of FIG. 1 having an inter-element delay T_1. After delaying the signal from first-order array **504** by an additional time delay T_2 at delay node **506**, the difference between the two first-order signals is generated at subtraction node **508** to generate the output signal $y_2(t)$ of ADMA **500**.

When farfield conditions apply, the magnitude of the frequency and angular dependent response $H_2(f, \theta)$ of second-order ADMA **500** is given by Equation (8) as follows:

$$|H_2(f,\theta)| = \left|\frac{Y_2(f,\theta)}{S(f)}\right| = 4\prod_{v=1}^{2} \sin\frac{2\pi f[T_v + (d_v\cos\theta)/c]}{2}, \tag{8}$$

where $Y_2(f, \theta)$ is the spectrum of the ADMA output signal $y_2(t)$. For the special case of small spacing and delay, i.e., kd_1, $kd_2 \ll \pi$ and T_1, $T_2 \ll \frac{1}{2}f$, Equation (8) may be written as Equation (9) as follows:

$$\left|H_2(f,\theta)\right| \approx (2\pi f)^2 \prod_{v=1}^{2} [T_v + (d_v\cos\theta)/c]. \tag{9}$$

Analogous to the case of first-order differential array **200** of FIG. 2, the amplitude response of second-order array **500** consists of a monopole term, a dipole term (cos θ), and an additional quadrapole term ($\cos^2\theta$). Also, a quadratic rise as a function of frequency can be observed. This frequency dependence can be equalized by applying a second-order lowpass filter. The directivity response can then be expressed by Equation (10) as follows:

$$D_2(\theta) = \prod_{v=1}^{2}\left(\frac{T_v}{T_v + d_v/c} + \left(1 - \frac{T_v}{T_v + d_v/c}\right)\cos\theta\right), \tag{10}$$

which is a direct result of the pattern multiplication theorem in electroacoustics.

One design goal of a second-order differential farfield array, such as ADMA **500** of FIG. **5**, may be to use the array in a host-based environment without the need for any special purpose hardware, e.g., additional external DSP interface boards. Therefore, two dipole elements may be utilized in order to form the second-order array instead of four omnidirectional elements. As a consequence, $T_1=0$ which means that one null angle is fixed to $\theta_{21}=90°$. In this case, although two independent nulls can be formed by the second-order differential array, only one can be made adaptive if two dipole elements are used instead of four omnidirectional transducers. The implementation of such a second-order ADMA may be based on first-order cardioid ADMA **200** of FIG. 2, where $d=d_2$, $T=T_2$, $\beta=\beta_2$, and d_1 is the acoustical dipole length of the dipole transducer. Additionally, the lowpass filter is chosen to be a second-order lowpass filter. FIG. **6** shows the directivity pattern of such a second-order back-to-back cardioid system. Those skilled in the art will understand that a second-order ADMA can also be implemented with three omnidirectional elements.

The transfer function $H_2(f, \theta)$ of a second-order ADMA formed of two dipole elements can be written according to Equation (11) as follows:

$$H_2(f,\theta) = \frac{Y_{out}(f,\theta)}{S(f)} = -4e^{-j\pi f T_2}\sin\left(\frac{kd_1\cos\theta}{2}\right)\cdot\left(\sin\frac{kd_2(1+\cos\theta)}{2} - \beta_2\sin\frac{kd_2(1-\cos\theta)}{2}\right), \tag{11}$$

with null angles given by Equations (12a) and (12b) as follows:

$$\theta_{21}=90°, \tag{12a}$$

$$\theta_{22} \approx \arccos\frac{\beta_2 - 1}{\beta_2 + 1}, \tag{12b}$$

where $0 \leqq \beta_2 \leqq 1$ under the constraint $90° \leqq \beta_{22}23$ 180°. FIG. **7** shows the directivity patterns that can be obtained by a second-order ADMA formed from two dipole elements for θ_{22} values of 90°, 120°, 150°, and 180°.

As shown in Elko, G. W., "Superdirectional Microphone Arrays," *Acoustic Signal Processing for Telecommunication*, J. Benesty and S. L. Gay (eds.), pp. 181–236, Kluwer Academic Publishers, 2000, a second-order differential array is typically superior to a first-order differential array in terms of directivity index, front-to-back ratio, and beamwidth.

Subband ADMA

FIG. **8** shows a schematic diagram of a subband two-element ADMA **800** comprising two elements **802** and **804**. When elements **802** and **804** are omnidirectional elements, ADMA **800** is a first-order system; when elements **802** and **804** are dipole elements, ADMA **800** is a second-order system. ADMA **800** is analogous to fullband ADMA **200** of FIG. **2**, except that one additional degree of freedom is obtained for ADMA **800** by performing the adaptive algorithm independently in different frequency subbands. In particular, delay nodes **806** and **808** of subband ADMA **800** are analogous to delay nodes **206** and **208** of fullband ADMA **200**; subtraction nodes **810**, **812**, and **816** of ADMA **800** are analogous to subtraction nodes **210**, **212**, and **216** of ADMA **200**; amplifier **814** of ADMA **800** is analogous to amplifier **214** of ADMA **200**; and lowpass filter **818** of ADMA **800** is analogous to lowpass filter **218** of ADMA **200**, except that, for ADMA **800**, the processing is independent for different frequency subbands.

To provide subband processing, analysis filter banks **820** and **822** divide the electrical signals from elements **802** and **804**, respectively, into two or more subbands l, and amplifier **814** can apply a different gain β(l,i) to each different subband l in the backward-facing cardioid signal $c_B(l,i)$. In addition, synthesis filter bank **824** combines the different subband signals y(l,i) generated at summation node **816** into a single fullband signal y(t), which is then lowpass filtered by filter **818** to generate the output signal $y_{out}(t)$ of ADMA **800**.

The gain parameter β(l,i), where l denotes the subband bin and i is the discrete time instance, is preferably updated by an adaptive algorithm that minimizes the output power of the array. This update therefore effectively adjusts the response of the backward-facing cardioid $c_B(l,i)$ and can be written according to Equations (13a) and (13b) as follows;

$$y(l,i)=c_F(l,i)-\beta(l,i)c_B(l,i), \tag{13a}$$

$$\bar{\beta}(l, i+1) = \beta(l, i) + \frac{\mu y(l, i) c_B(l, i)}{\|c_B(l, i)\|^2 + a}, \quad (13b)$$

where

$$\beta(l, i) = \begin{cases} \bar{\beta}(l, i), & 0 \le \bar{\beta}(l, i) \le 1 \\ 0, & \bar{\beta}(l, i) < 0 \\ 1, & \bar{\beta}(l, i) > 1 \end{cases}, \quad (14)$$

and μ is the update parameter and α is a positive constant.

By using this algorithm, multiple spatially distinct noise sources with non-overlapping spectra located in the back half plane of the ADMA can be tracked and attenuated simultaneously.

Implementation and Measurements

PC-based real-time implementations running under the Microsoft™ Windows™ operating system were realized using a standard soundcard as the analog-to-digital converter. For these implementations, the demonstrator's analog front-end comprised two omnidirectional elements of the type Panasonic WM-54B as well as two dipole elements of the type Panasonic WM-55D103 and a microphone preamplifier offering 40-dB gain comprise the analog front-end. The implementations of the first-order ADMAs of FIGS. **2** and **8** utilized the two omnidirectional elements and the preamplifier, while the implementation of the second-order ADMA of FIG. **5** utilized the two dipole elements and the preamplifier.

The signals for the forward-facing cardioids $c_F(t)$ and the backward-facing cardioids $c_B(t)$ of the first-order ADMAs of FIGS. **2** and **8** were obtained by choosing the spacing d between the omnidirectional microphones such that there is one sample delay between the corresponding delayed and undelayed microphone signals. Similarly, the signals for the forward- and backward-facing cardioids of the second-order ADMA of FIG. **5** were obtained by choosing the spacing d_2 between the dipole microphones such that there is one sample delay between the corresponding delayed and undelayed microphone signals. Thus, for example, for a sampling frequency f_s of 22050 Hz, the microphone spacing $d = d_2 =$ 1.54 cm. For the Panasonic dipole elements, the acoustical dipole length d_1 was found to be 0.8 cm.

FIGS. **9**A and **9**B depict the fullband ADMA directivity patterns for first-order and second-order arrays, respectively. These measurements were performed by placing a broadband jammer (noise source) at approximately 90° with respect to the array's axis (i.e., θ_1 for the first-order array and θ_{22} for the second-order array) utilizing a standard directivity measurement technique. It can be seen that deep nulls covering wide frequency ranges are formed in the direction of the jammer.

FIGS. **10** and **11** show measured directivity of first- and second-order subband implementations of ADMA **800** of FIG. **8**, respectively, for four simultaneously playing sinusoids. For the first-order subband implementation, four loudspeakers simultaneously played sinusoidal signals while positioned in the back half plane of the arrays at θ_1 values of approximately 90°, 120°, 150°, and 180°. For the second-order subband implementation, four loudspeakers simultaneously played sinusoidal signals while positioned in the back half plane of the arrays at θ_{22} values of approximately 110°, 120°, 150°, and 180°. As can be seen, these measurements are in close agreement with the simulated patterns shown in FIGS. **4** and **7**.

In order to combat the noise amplification properties inherent in differential arrays, the demonstrator included a noise reduction method as presented in Diethorn, E. J., "A Subband Noise-Reduction Method for Enhancing Speech in Telephony & Teleconferencing," *IEEE Workshop on Applications of Signal Processing to Audio and Acoustics*, Mohonk, USA, 1997, the teachings of which are incorporated herein by reference.

Conclusions

First- and second-order ADMAs which are able to adaptively track and attenuate a possibly moving noise source located in the back half plane of the arrays have been presented. It has been shown that, by performing the calculations in subbands, even multiple spatially distinct noise sources with non-overlapping spectra can be tracked and attenuated simultaneously. The real-time implementation presents the dynamic performance of the ADMAs in real acoustic environments and shows the practicability of using these arrays as acoustic front-ends for a variety of applications including telephony, automatic speech recognition, and teleconferencing.

The present invention may be implemented as circuit-based processes, including possible implementation on a single integrated circuit. As would be apparent to one skilled in the art, various functions of circuit elements may also be implemented as processing steps in a software program. Such software may be employed in, for example, a digital signal processor, micro-controller, or general-purpose computer.

The present invention can be embodied in the form of methods and apparatuses for practicing those methods. The present invention can also be embodied in the form of program code embodied in tangible media, such as floppy diskettes, CD-ROMs, hard drives, or any other machine-readable storage medium, wherein, when the program code is loaded into and executed by a machine, such as a computer, the machine becomes an apparatus for practicing the invention. The present invention can also be embodied in the form of program code, for example, whether stored in a storage medium, loaded into and/or executed by a machine, or transmitted over some transmission medium or carrier, such as over electrical wiring or cabling, through fiber optics, or via electromagnetic radiation, wherein, when the program code is loaded into and executed by a machine, such as a computer, the machine becomes an apparatus for practicing the invention. When implemented on a general-purpose processor, the program code segments combine with the processor to provide a unique device that operates analogously to specific logic circuits.

The use of figure reference labels in the claims is intended to identify one or more possible embodiments of the claimed subject matter in order to facilitate the interpretation of the claims. Such labeling is not to be construed as necessarily limiting the scope of those claims to the embodiments shown in the corresponding figures.

It will be further understood that various changes in the details, materials, and arrangements of the parts which have been described and illustrated in order to explain the nature of this invention may be made by those skilled in the art without departing from the scope of the invention as expressed in the following claims.

What is claimed is:

1. A second-order adaptive differential microphone array (ADMA), comprising:

(a) a first first-order element configured to convert a received audio signal into a first electrical signal;

(b) a second first-order element configured to convert the received audio signal into a second electrical signal;

(c) a first delay node configured to delay the first electrical signal from the first first-order element to generate a delayed first electrical signal;

(d) a second delay node configured to delay the second electrical signal from the second first-order element to generate a delayed second electrical signal;

(e) a first subtraction node configured to generate a forward-facing cardioid signal based on a difference between the first electrical signal and the delayed second electrical signal;

(f) a second subtraction node configured to generate a backward-facing cardioid signal based on a difference between the second electrical signal and the delayed first electrical signal;

(g) an amplifier configured to amplify the backward-facing cardioid signal by a gain parameter to generate an amplified backward-facing cardioid signal; and

(h) a third subtraction node configured to generate a difference signal for the second-order ADMA based on a difference between the forward-facing cardioid signal and the amplified backward-facing cardioid signal.

2. The invention of claim 1, further comprising a lowpass filter configured to filter the difference signal from the third subtraction node to generate an output signal for the second-order ADMA.

3. The invention of claim 1, wherein the first and second first-order elements are two dipole elements.

4. The invention of claim 1, wherein each of the first and second first-order elements is a first-order differential microphone array.

5. The invention of claim 4, wherein each first-order differential microphone array comprises:

(1) a first omnidirectional element configured to convert the received audio signal into an electrical signal;

(2) a second omnidirectional element configured to convert the received audio signal into an electrical signal;

(3) a delay node configured to delay the electrical signal from the second omnidirectional element to generate a delayed electrical signal; and

(4) a first subtraction node configured to generate the corresponding electrical signal for the first-order element based on a difference between the electrical signal from the first omnidirectional element and the delayed electrical signal from the delay node.

6. The invention of claim 1, wherein the gain parameter for the amplifier is configured to be adaptively adjusted to move a null located in a back half plane of the second-order ADMA to track a moving noise source.

7. The invention of claim 6, wherein the gain parameter is configured to be adaptively adjusted to minimize output power from the second-order ADMA.

8. The invention of claim 1, further comprising:

(i) a first analysis filter bank configured to divide the first electrical signal from the first first-order element into two or more subband electrical signals corresponding to two or more different frequency subbands;

(j) a second analysis filter bank configured to divide the second electrical signal from the second first-order element into two or more subband electrical signals corresponding to the two or more different frequency subbands; and

(k) a synthesis filter bank configured to combine two or more different subband difference signals generated by the third difference node to form a fullband difference signal.

9. The invention of claim 8, wherein the amplifier is configured to apply a different subband gain parameter to a backward-facing subband cardioid signal generated by the second subtraction node for each different frequency subband.

10. The invention of claim 9, wherein each different subband gain parameter is configured to be adaptively adjusted to move a different null in a back half plane of the second-order ADMA to track a different moving noise source corresponding to each different frequency subband.

11. The invention of claim 10, wherein each different subband gain parameter is configured to be adaptively adjusted to minimize output power from the second-order ADMA in the corresponding frequency subband.

12. An apparatus for processing signals generated by a microphone array (ADMA) having (i) a first first-order element configured to convert a received audio signal into a first electrical signal and (ii) a second first-order element configured to convert the received audio signal into a second electrical signal, the apparatus comprising:

(a) a first delay node configured to delay the first electrical signal from the first first-order element to generate a delayed first electrical signal;

(b) a second delay node configured to delay the second electrical signal from the second first-order element to generate a delayed second electrical signal;

(c) a first subtraction node configured to generate a forward-facing cardioid signal based on a difference between the first electrical signal and the delayed second electrical signal;

(d) a second subtraction node configured to generate a backward-facing cardioid signal based on a difference between the second electrical signal and the delayed first electrical signal;

(e) an amplifier configured to amplify the backward-facing cardioid signal by a gain parameter to generate an amplified backward-facing cardioid signal; and

(f) a third subtraction node configured to generate a difference signal for the second-order ADMA based on a difference between the forward-facing cardioid signal and the amplified backward-facing cardioid signal.

13. The invention of claim 12, further comprising a lowpass filter configured to filter the difference signal from the third subtraction node to generate an output signal for the second-order ADMA.

14. The invention of claim 12, wherein the first and second first-order elements are two dipole elements.

15. The invention of claim 12, wherein each of the first and second first-order elements is a first-order differential microphone array.

16. The invention of claim 15, wherein each first-order differential microphone array comprises:

(1) a first omnidirectional element configured to convert the received audio signal into an electrical signal;

(2) a second omnidirectional element configured to convert the received audio signal into an electrical signal;

(3) a delay node configured to delay the electrical signal from the second omnidirectional element to generate a delayed electrical signal; and

(4) a first subtraction node configured to generate the corresponding electrical signal for the first-order element based on a difference between the electrical signal

from the first omnidirectional element and the delayed electrical signal from the delay node.

17. The invention of claim **12**, wherein the gain parameter for the amplifier is configured to be adaptively adjusted to move a null located in a back half plane of the second-order ADMA to track a moving noise source.

18. The invention of claim **17**, wherein the gain parameter is configured to be adaptively adjusted to minimize output power from the second-order ADMA.

19. The invention of claim **12**, further comprising:

(g) a first analysis filter bank configured to divide the first electrical signal from the first first-order element into two or more subband electrical signals corresponding to two or more different frequency subbands;

(h) a second analysis filter bank configured to divide the second electrical signal from the second first-order element into two or more subband electrical signals corresponding to the two or more different frequency subbands; and

(i) a synthesis filter bank configured to combine two or more different subband difference signals generated by the third difference node to form a fullband difference signal.

20. The invention of claim **19**, wherein the amplifier is configured to apply a different subband gain parameter to a backward-facing subband cardioid signal generated by the second subtraction node for each different frequency subband.

21. The invention of claim **20**, wherein each different subband gain parameter is configured to be adaptively adjusted to move a different null in a back half plane of the second-order ADMA to track a different moving noise source corresponding to each different frequency subband.

22. The invention of claim **21**, wherein each different subband gain parameter is configured to be adaptively adjusted to minimize output power from the second-order ADMA in the corresponding frequency subband.

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